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ABSTRACT:

CHG DATE=19990617 STATUS=O> A circuit arrangement for automatically adapting the volume of a loudspeaker to an interfering noise level prevailing at the loudspeaker location is specified, particularly for mobile radio receivers such as car radios or the like, which exhibits an adjustable function network (18)

with a transfer function, which generates a correcting variable for a volume controller (11) in dependence on the interfering noise level in such a manner that the useful voltage level at the loudspeaker is logically combined with the interfering noise level via a predetermined characteristic. For the purpose of adapting the volume to the interfering noise level in accordance with an individual characteristic which can be automatically adapted to the situation in an internal vehicle space and to the requirements and the auditory sensation of the motor vehicle passengers, a curve adaptor (19) is connected to the function network (18) and a volume control (12), which adjusts the function network (18) in such a manner that the predetermined characteristic can be modified at least section by section in dependence on the operation of the volume control (Figure 1). <IMAGE>

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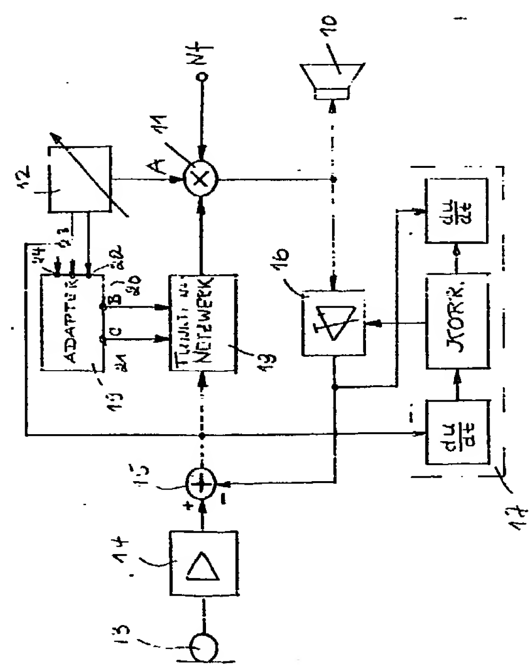
⑤④ **Schaltungsanordnung zum selbsttätigen Anpassen der Lautstärke eines Lautsprechers an einen am Lautsprecherort herrschenden Störgeräuschpegel**

Es wird eine Schaltungsanordnung zum selbsttätigen Anpassen der Lautstärke eines Lautsprechers an einen am Lautsprecherort herrschenden Störgeräuschpegel, insbesondere für mobile Rundfunkempfänger, wie Autoradios od. dgl., angegeben, der ein einstellbares Funktionsnetzwerk (18) mit Übertragungsfunktion aufweist, das eine Stellgröße für einen Lautstärkeregler (11) in Abhängigkeit von dem Störgeräuschpegel derart generiert, daß der Nutzspannungspegel am Lautsprecher mit dem Störgeräuschpegel über eine vorgegebene Kennlinie verknüpft ist. Zum Zwecke der Anpassung der Lautstärke an den Störgeräuschpegel nach einer individuellen, an die Gegebenheiten eines Kraftfahrzeuginnenraums und an die Bedürfnisse und das Hörempfinden des Kraftfahrzeuginsassen automatisch adaptierbaren Kennlinie, ist mit dem Funktionsnetzwerk (18) und einem Lautstärkesteller (12) ein Kennlinienadapter (19) verbunden, der das Funktionsnetzwerk (18) derart einstellt, daß die vorgegebene Kennlinie in Abhängigkeit von der Betätigung des Lautstärkestellers zumindest abschnittweise modifizierbar ist (Fig. 1).



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Patentansprüche

1. Schaltungsanordnung zum selbsttätigen Anpassen der Lautstärke eines Lautsprechers an einen am Lautsprecherort herrschenden Störgeräuschpegel, insbesondere für mobile Rundfunkempfänger, wie Autoradios od.dgl., mit einem dem Lautsprecher zugeordneten Lautstärkeregler mit darauf einwirkendem manuellen Lautstärkesteller, mit einem zugleich das Störgeräusch und das Lautsprechersignal erfassenden Mikrofon, mit einer die jeweils gleichgerichtete und/oder quadrierte Mikrofonausgangsspannung und Lautsprechereingangsspannung mit entgegengesetzter Polarität verknüpfenden Summierschaltung, aus deren Ausgangsspannung eine dem Lautstärkeregler zugeführte Stellgröße abgeleitet ist, und mit einem im Verbindungszweig von Lautsprechereingang und Summierschaltungseingang angeordneten Spannungsverstärker mit einstellbarer Spannungsverstärkung, d a - d u r c h g e k e n n z e i c h n e t, daß die Ausgangsspannung der Summierschaltung (15) einem zwischen Summierschaltung (15) und Lautstärkeregler (11) eingeschalteten Funktionsnetzwerk (18) mit einstellbarer Übertragungsfunktion zugeführt ist, der die Stellgröße derart generiert, daß die Nutzspannung (U_{Nutz}) am Ausgang des Lautstärkereglers (11) über eine vorgegebene Stellkennlinie mit der Ausgangs-

spannung ($U_{Stör}$) der Summierschaltung (15) verknüpft ist, und daß mit dem Lautstärkesteller (12) und dem Funktionsnetzwerk (18) ein das Funktionsnetzwerk (18) derart einstellender Kennlinienadapter (22) verbunden ist, daß die vorgegebene Stellkennlinie in Abhängigkeit von der Betätigung des Lautstärkestellers (12) zumindest abschnittweise modifizierbar ist.

2. Schaltungsanordnung nach Anspruch 1, dadurch gekennzeichnet, daß die Stellkennlinie in drei Kennlinienabschnitte (a,b,c) unterteilt ist, daß in einem ersten Abschnitt (a) der Nutzspannungspegel (U_{Nutz}) am Ausgang des Lautstärkereglers (11) einen von der Einstellung des Lautstärkestellers (12) abhängigen Betrag aufweist, daß in einem zweiten Abschnitt (b) der Nutzspannungspegel (U_{Nutz}) proportional dem Spannungspegel ($U_{Stör}$) am Ausgang der Summierschaltung (15) ist, daß in einem dritten Abschnitt (c) der Nutzspannungspegel (U_{Nutz}) auf einen maximalen Betrag begrenzt ist und daß die drei Kennlinienabschnitte (a,b,c) mit von Null an zunehmendem Spannungspegel ($U_{Stör}$) am Ausgang der Summierschaltung (15) nacheinander durchlaufen werden.

3. Schaltungsanordnung nach Anspruch 2, dadurch gekennzeichnet, daß die Zuordnung der Kennlinienabschnitte (a,b,c) zu dem Nutzspannungspegel (U_{Nutz}) und zu dem Spannungspegel ($U_{Stör}$) am Ausgang der Summierschaltung (15) jeweils logarithmisch vorgenommen ist.

4. Schaltungsanordnung nach Anspruch 3, d a - d u r c h g e k e n n z e i c h n e t, daß der Kennlinienadapter (19) derart ausgebildet ist, daß er bei am Ausgang der Summierschaltung (15) anstehenden Spannungspegelwerten ($U_{Stör}$), die dem zweiten Kennlinienabschnitt (b) zugehörig sind, das Funktionsnetzwerk (18) so einstellt, daß sich die vorgegebene Stellkennlinie insgesamt zu kleineren Spannungspegelwerten ($U_{Stör}$) verschiebt, wenn der Lautstärkesteller (12) häufiger zur Lautstärkeerhöhung als zur Lautstärkereduzierung verstellt wird, und zu größeren Spannungspegelwerten ($U_{Stör}$) verschiebt, wenn der Lautstärkesteller (12) häufiger zur Lautstärkereduzierung als zur Lautstärkeerhöhung verstellt wird.

5. Schaltungsanordnung nach Anspruch 3 oder 4, d a d u r c h g e k e n n z e i c h n e t, daß der Kennlinienadapter (19) derart ausgebildet ist, daß er bei am Ausgang der Summierschaltung (15) anstehenden Spannungspegelwerten ($U_{Stör}$), die dem dritten Kennlinienabschnitt (c) zugehörig sind, das Funktionsnetzwerk(18) derart einstellt, daß der Maximalbetrag des Nutzspannungspegels (U_{Nutz}) am Ausgang des Lautstärkereglers (11) vergrößert wird, wenn der Lautstärkesteller (12) häufiger zur Lautstärkeerhöhung als zur Lautstärkereduzierung verstellt wird, und verkleinert wird, wenn der Lautstärkesteller (12) häufiger zur Lautstärkereduzierung als zur Lautstärkeerhöhung verstellt wird.

6. Schaltungsanordnung nach einem der Ansprüche 1 - 5, d a d u r c h g e k e n n z e i c h n e t, daß der Kennlinienadapter (19) einen Schwellwertdiskriminator (25) mit zwei Ausgängen (26,27), an dessen

einem Ausgang (26) ein Signal ansteht, wenn das Eingangssignal zwischen einem ersten und zweiten Schwellwert liegt, und an dessen anderem Ausgang ein Signal ansteht, wenn das Eingangssignal den zweiten Schwellwert übersteigt, und zwei Vorwärts-Rückwärts-Zähler (32,33), deren Zählinhalte jeweils als Steuergrößen (B,C) für das Funktionsnetzwerk (18) am Ausgang (20,21) des Kennlinienadapters (19) anliegen, aufweist, daß der Eingang (24) des Schwellwertdiskriminators (25) mit dem Ausgang der Summierschaltung (15) verbunden ist und die Ausgänge (26,27) des Schwellwertdiskriminators (25) über jeweils ein Torglied (30,31) an jeweils einem Zähleingang der Vorwärts-Rückwärts-Zähler (32,33) angeschlossen sind, daß der Lautstärkeregler (12) derart ausgebildet ist, daß mit jeder Verstellung ein die Verstellrichtung (laut/leise) charakterisierendes Steuersignal und mindestens ein Zählimpuls ausgegeben werden, und daß das Steuersignal an dem Zählrichtungseingang der Vorwärts-Rückwärts-Zähler (32,33) und der Zählimpuls am Steuereingang der Torglieder (30,31) liegt.

7. Schaltungsanordnung nach Anspruch 7, d a d u r c h g e k e n n z e i c h n e t, daß der Schwellwertdiskriminator (25) von einem Fensterkomparator (28) und einem Komparator (29) gebildet sind.

8. Schaltungsanordnung nach einem der Ansprüche 1 - 7, d a d u r c h g e k e n n z e i c h n e t, daß der Steuereingang des Spannungsverstärkers (16) mit einem Abgleichkreis (17) verbunden ist, der

die Spannungsverstärkung laufend derart nachstellt, daß die in der Mikrofonausgangsspannung enthaltene, vom Lautsprechersignal herrührende Nutzspannungskomponente in der Summierschaltung (15) vollständig kompensiert wird.

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Schaltungsanordnung zum selbsttätigen Anpassen der Lautstärke eines Lautsprechers an einen am Lautsprecherort herrschenden Störgeräuschpegel

Die Erfindung betrifft eine Schaltungsanordnung zum selbsttätigen Anpassen der Lautstärke eines Lautsprechers an einen am Lautsprecherort herrschenden Störgeräuschpegel, insbesondere für mobile Rundfunkempfänger, wie Autoradios od.dgl., der im Oberbegriff des Anspruchs 1 definierten Gattung.

Mit einer solchen Schaltungsanordnung wird die Lautstärke entsprechend dem Geräuschpegel in der Umgebung des Lautsprechers so eingestellt, daß der Wiedergabepegel des Lautsprechers, also der Nutzsignalpegel immer um einige Dezibel (dB) höher liegt als der Störgeräuschpegel. Dadurch wird das vom Lautsprecher abgestrahlte Nutzsignal von dem Hörenden, unabhängig von dem jeweiligen Stärkegrad der Umweltgeräusche, in etwa immer gleich laut empfunden. Insbesondere für den mobilen Betrieb des beispielsweise mit einem Rundfunkempfänger verbundenen Lautsprechers bedeutet dies eine wesentliche Verbesserung des Bedienungskomforts, da der Hörende nicht mehr gezwungen ist, bei häufig wechselndem Störgeräuschpegel die Lautstärke des Lautsprechers

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ständig nachzustellen.

Um das im Mikrofon aufgenommene Nutzsignal zu eliminieren, ist es erforderlich, den Kopplungsfaktor zwischen Mikrofon und Lautsprecher zu kennen. Dieser ist jedoch unter anderem abhängig von Fahrzeug, dem Einbauort von Lautsprecher und Mikrofon und Lautsprecherart.

Bei einer bekannten Schaltungsanordnung dieser Art (DE-OS 29 42 331) ist daher dem durch einen Gleichrichter und ein Quadriernetzwerk realisierten Gegenspannungsverstärker ein Abstimmglied zugeordnet, mit welchem die Spannungsverstärkung nach Einbau der Schaltungsanordnung in das jeweilige Kraftfahrzeug solange verändert werden kann, bis der Einfluß des Lautsprechersignals in der dem Lautstärkeregler zugeführten, von der Summierschaltung ausgegebenen Stellgröße vollständig kompensiert ist. Die richtige Abstimmung der Schaltungsanordnung läßt sich daran erkennen, daß ein Einfluß des Lautsprechersignals auf die Lautstärkeeinstellung nicht mehr wahrgenommen werden kann.

Die für den Kraftfahrzeuginsassen subjektiv richtige Lautstärkeanhebung bei Auftreten von Störgeräuschen ist aber auch von den Kopplungsfaktoren zwischen dem menschlichen Ohr einerseits und dem Störgeräusch und dem Nutzsignal andererseits abhängig. Diese Kopplungsfaktoren ändern sich ebenfalls je nach Fahrzeugart, Einbauort von Lautsprecher und Mikrofon, durch die Kraftfahrzeug-Besetzung oder durch Veränderung der Fader- und Balance-Stellung. Diese Kopplungsfaktoren sind individueller Art und können nur über das menschliche Ohr selbst ermittelt werden.

Der Erfindung liegt die Aufgabe zugrunde, eine Schaltungsanordnung der eingangs genannten Art zu schaffen, welche die Anpassung der Lautstärke an den Störgeräuschpegel nach einer individuell an die Gegebenheiten eines Kraftfahrzeug-Innenraums und an die Bedürfnisse und das Hörempfinden des oder der Kraftfahrzeuginsassen automatisch anpaßbaren Nutzsignal-Störgeräusch-Zuordnung, der sogenannten Stellkennlinie, durchführt.

Die Aufgabe ist bei einer Schaltungsanordnung der im Oberbegriff des Anspruchs 1 definierten Gattung erfindungsgemäß durch die Merkmale im Kennzeichnungsteil des Anspruchs 1 gelöst.

Die erfindungsgemäße Schaltungsanordnung hat den Vorteil, daß eine werksseitig vorgegebene und auf ein Standardfahrzeug und ein Normhörempfinden zugeschnittene Störpegel-Nutzpegel-Stellkennlinie mit vorgegebenem Grundlautstärkepegel, Endlautstärkepegel und vorgegebener Kennliniensteilheit "selbstlernend" in einer solchen Art automatisch modifiziert wird, daß die Stellkennlinie nach dem "Lernprozeß" optimal an die räumlichen Gegebenheiten des Kraftfahrzeuginnenraumes und an das Hörempfinden des Benutzers adaptiert ist. Bei diesem "Lernprozeß" dient der Lautstärkesteller als Rezeptor, der die Reaktion des Benutzers auf die Lautstärkeanpassung, die sich in der Betätigung des Lautstärkestellers äußert, aufnimmt und als Maß für die Kennlinienadaption der Schaltungsanordnung verfügbar macht.

Vorteilhafte Ausführungsformen der Erfindung sind den weiteren Ansprüchen 2 - 8 zu entnehmen. Insbesondere in Verbindung mit der Ausführungsform gemäß Anspruch 8, mit welcher sich ein automatischer Abgleich der Schaltungsanordnung in Hinblick auf die vollständige Kompensation der Nutzsignalkomponente im Mikrofonausgangssignal nach Einbau im Kraftfahrzeug erreichen läßt, wird eine Schaltungsanordnung zur störgeräuschabhängigen Lautstärkeeinstellung erzielt, die an alle räumlichen und subjektiven Gegebenheiten optimal angepaßt ist und keine Wünsche offenläßt.

Die Erfindung ist anhand eines in der Zeichnung dargestellten Ausführungsbeispiels im folgenden näher beschrieben. Es zeigen:

Fig. 1 ein Blockschaltbild einer Schaltungsanordnung zur störgeräuschabhängigen Lautstärkeeinstellung für ein Autoradio,

Fig. 2 ein Diagramm einer Stellkennlinie, welches den gewünschten Verlauf des Nutzsignalpegels in Abhängigkeit von einem Störgeräuschpegel darstellt,

Fig. 3 ein Blockschaltbild eines Kennlinienadapters der Schaltungsanordnung in Fig. 1.

In der in Fig. 1 dargestellten Schaltungsanordnung ist mit 10 der Lautsprecher eines Autoradios und mit 11 ein im Niederfrequenzteil des Rundfunkempfängers angeordneter Lautstärkeregler bezeichnet. Der Lautstärkeregler 11

ist hier als Multiplizierer ausgebildet, dem einerseits das Niederfrequenzsignal Nf des Rundfunkempfängers und andererseits die Stellgröße eines manuell zu bedienenden Lautstärkestellers 12 zugeführt ist. Der Lautstärkeregler 11 kann mit dem Lautstärkesteller 12 zu einem integrierten Verstärker zusammengefaßt sein, wie er in der DE-OS 29 04 920 (Fig. 2) beschrieben ist.

Die Schaltungsanordnung weist ferner ein Mikrofon 13 auf, das sowohl das Lautsprechersignal als auch das Störgeräusch im Fahrgastraum des Autos erfaßt. Das Mikrofon 13 ist entweder ebenso wie die Schaltungsanordnung im Radiogehäuse oder in dessen Nähe angeordnet. Entsprechend dem aufgenommenen Störgeräusch und Lautsprechersignal erzeugt das Mikrofon 13 eine Ausgangsspannung, die sich aus einer vom Störgeräusch und von dem Lautsprechersignal herrührenden Spannungskomponente zusammensetzt. Der Ausgang des Mikrofons 13 ist über einen Mikrofonverstärker 14 mit Gleichrichter, in welchem vorzugsweise auch eine Quadrierung der Eingangsspannung vorgenommen wird, mit einer Summierschaltung 15 verbunden. Der zweite, negierte Eingang der Summierschaltung 15 ist mit dem Ausgang eines steuerbaren Spannungsverstärkers 16 verbunden, der eingangsseitig mit dem Eingang des Lautsprechers 10 bzw. dem Ausgang des Lautstärkereglers 11 verbunden ist. Im Spannungsverstärker 16 wird ebenfalls eine Gleichrichtung und Quadrierung der Eingangsspannung, also der Nutzsignalspannung U_{Nutz}, durchgeführt.

In der Summierschaltung 15 werden die beiden Eingangsspannungen, also die verstärkte Mikrofonausgangsspannung und die verstärkte Lautsprechereingangsspannung,

gegensinnig addiert. Aus der am Ausgang der Summierschaltung 15 anstehenden Differenzspannung wird eine Stellgröße für den Lautstärkeregler 11 abgeleitet, die den Lautstärkeregler 11 entsprechend einstellt, im vorliegenden Fall eines Multiplizierers die Nutzspannung im Niederfrequenzteil mit einem entsprechenden Faktor multipliziert.

Die Schaltungsanordnung muß nunmehr derart abgestimmt oder abgeglichen sein, daß die Differenzspannung am Ausgang der Summierschaltung 15 nur noch die vom Störgeräusch herrührende Spannungskomponente enthält, die in der Mikrofonausgangsspannung enthaltene, vom Nutzsignal herrührende Nutzspannungskomponente also durch die Gegenspannung des Spannungsverstärkers 16 vollständig kompensiert ist. Zum selbsttätigen Abstimmen oder Abgleich der Schaltungsanordnung ist ein Abgleichkreis 17 vorgesehen, der ausgangsseitig an dem Steuereingang des Spannungsverstärkers 16 und eingangsseitig sowohl an dem Ausgang der Summierschaltung 15 als auch an dem Ausgang des Spannungsverstärkers 16 angeschlossen ist. Ein solcher Abgleichkreis 17 ist in der Patentanmeldung P 33 20 751.8 (Fig. 3) beschrieben, so daß hier lediglich kurz dessen Funktion in Erinnerung gerufen wird. Der aus zwei Differenzierglieder und einem Korrelator bestehende Abgleichkreis 17 erfaßt die Spannungsänderungen an den Ausgängen von Summierschaltung 15 und Spannungsverstärker 16. Ist die Gegenspannungsverstärkung des Spannungsverstärkers 16 zu niedrig eingestellt, so wächst die Ähnlichkeit der Ausgangsspannungen der Differenzierglieder zunehmend mit abnehmender Spannungsverstärkung. Im Extremfall (Ausgangspannung des Spannungsverstärkers 16 ist Null) sind die Ausgangsspannungen der Differenzierglieder am ähnlichsten und die Korrelationsfunktion weist ein Maximum auf. Entsprechend der Größe der Korrelations-

funktion gibt der Korrelator ein Stellsignal an den Spannungsverstärker 16. Die Spannungsverstärkung wird vergrößert. Ist hingegen die Spannungsverstärkung zu hoch eingestellt, so wächst zwar auch die Ähnlichkeit der Ausgangsspannungen der Differenzierglieder mit zunehmender Spannungsverstärkung, jedoch sind diese Ausgangsspannungen gegenphasig. Die Korrelationsfunktion wird immer kleiner und strebt schließlich gegen ein Minimum, wenn die vom Störgeräusch herrührende Spannungskomponente vernachlässigbar klein ist gegenüber der am Eingang der Summierschaltung 15 liegenden Nutzspannungskomponente. In diesem Fall verringerte sich die Ausgangsspannung des Korrelators und die Spannungsverstärkung des Spannungsverstärkers 16 wird reduziert. Bei abgeglichener Schaltungsanordnung korrelieren die Ausgangssignale der beiden Differenzierglieder nicht miteinander. Die Korrelationsfunktion ist im wesentlichen Null, und die Schaltungsanordnung ist abgeglichen. Der Abgleichkreis 17 kann aber auch wie in der vorgenannten Patentanmeldung zu Fig. 1 beschrieben ausgebildet sein.

Ein einstellbares Funktionsnetzwerk 18 ist eingangsseitig mit dem Ausgang der Summierschaltung 15 und ausgangsseitig mit dem Lautstärkeregler 11, im Falle des integrierten Verstärkers gemäß DE-OS 29 04 920 mit dessen Steuereingang, verbunden. Das Funktionsnetzwerk weist eine Übertragungsfunktion auf, mit deren Hilfe aus der Differenzspannung am Ausgang der Summierschaltung 15 eine Stellgröße für den Lautstärkeregler 11 derart generiert wird, daß die Nutzspannung U_{Nutz} am Ausgang des Lautstärkereglers 11 über eine Stellkennlinie, wie sie in Fig. 2 ausgezogen dargestellt ist, mit der Ausgangsspannung der Summierschaltung, die ja nur noch die

vom Störgeräusch herrührende Spannungskomponente $U_{Stör}$ enthält, verknüpft ist. Einstellbare Funktionsnetzwerke mit beliebig realisierbaren Übertragungsfunktionen sind bekannt. Über Steuereingänge ist das Funktionsnetzwerk 18 in seiner Übertragungsfunktion derart änderbar, daß mittels der generierten Stellgröße eine Modifizierung der Stellkennlinie möglich ist, wie dies in Fig. 2 strichliniert dargestellt ist. Hierzu sind die Steuereingänge mit jeweils einem von zwei Ausgängen 20,21 eines Kennlinienadapters 19 verbunden, dessen Eingänge 22 und 23 mit dem Lautstärkesteller 12 und dessen Eingang 24 mit dem Ausgang der Summierschaltung 15 verbunden ist.

In dem Funktionsnetzwerk 18 ist werksseitig eine solche Übertragungsfunktion realisiert, daß die am Ausgang des Funktionsnetzwerkes 18 in Abhängigkeit von der am Eingang anliegenden Differenzspannung erhaltene Stellgröße der Schaltungsanordnung eine solche Stellkennlinie verleiht, wie sie in Fig. 2 ausgezogen dargestellt ist und den Zusammenhang zwischen Nutzpegel und Störpegel im doppellogarithmischen Maßstab charakterisiert. Die Stellkennlinie weist drei Kennlinienabschnitte a,b und c auf, die jeweils durch Störpegelschwellen vorgegeben sind. In dem ersten Abschnitt a weist der Nutzpegel einen vom Störgeräusch unabhängigen Betrag, den Grundlautstärkepegel auf. Dieser Grundlautstärkepegel ist lediglich von der Einstellung des Lautstärkestellers abhängig. In dem zweiten Abschnitt b ist der Nutzpegel proportional dem Störpegel, wobei die Kennliniensteilheit den Störabstandsverlauf als Funktion des Störpegels angibt. In dem dritten Abschnitt c ist der Nutzpegel auf einen max. Betrag, dem Endlautstärkepegel, begrenzt. Diese drei Kennlinienabschnitte werden mit von Null an

zunehmendem Störpegel nacheinander durchlaufen.

Der Kennlinienadapter 19 weist einen Schwellwertdiskriminator 25 mit zwei Ausgängen 26, 27 und einem Eingang auf, der mit dem Eingang 24 des Kennlinienadapters 19 verbunden ist. Der Schwellwertdiskriminator 25 ist derart aufgebaut, daß ein Ausgangssignal ausschließlich an dem Ausgang 26 ansteht, wenn das Eingangssignal zwischen einem ersten und zweiten Schwellwert liegt und ausschließlich an dem Ausgang 27 ansteht, wenn das Eingangssignal den zweiten Schwellwert übersteigt. Im vorliegenden Ausführungsbeispiel besteht der Schwellwertdiskriminator 25 aus einem Fensterkomparator 28 und einen Komparator 29, die eingangsseitig an dem Eingang 24 angeschlossen sind. Die beiden Ausgänge 26 und 27 des Schwellwertdiskriminators 25 sind über je ein als AND-Gatter ausgebildetes Torglied 30, 31 mit jeweils einem Vorwärts-Rückwärts-Zähler 32 bzw. 33 verbunden. Die Ausgänge der Vorwärts-Rückwärts-Zähler 32, 33 bilden die Ausgänge 20, 21 des Kennlinienadapters 19.

Der Lautstärkeregler 11 ist derart ausgebildet, daß mit jeder Verstellung sowohl ein die Verstellrichtung (laut/leise) charakterisierendes Steuersignal an dem Eingang 22 als auch ein Zählimpuls an den Eingang 23 des Kennlinienadapters 19 gelangt. An dem Eingang 22 sind die Zählrichtungseingänge "up/down" und an dem Eingang 23 sind die Steuereingänge der Torglieder 30, 31 bzw. die anderen Eingänge der sie bildenden AND-Gatter angeschlossen.

Liegt der momentane Störgeräuschpegel in dem Kennlinienabschnitt b, so übersteigt die Differenzspannung am Ausgang der Summierschaltung 15 den unteren Schwellwert des Schwellwertdiskriminators 25. Am Ausgang 26 steht ein Sig-

nal an, der das AND-Gatter 30 setzt. Wird nunmehr der Lautstärkesteller 13 betätigt, so gelangt ein Zählimpuls durch das gesetzte AND-Gatter 30 zu dem Takteingang des Vorwärts-Rückwärts-Zählers 32. Je nach Verstellrichtung des Lautstärkestellers 12 (laut oder leise) wird der Zählinhalt des Zählers 32 vergrößert oder verringert. Der Zählinhalt des Zählers 32 bildet die Stellgröße B, die das Funktionsnetzwerk 18 nunmehr so einstellt, daß sich die Stellkennlinie im Kennlinienabschnitt b in Richtung des Pfeils B verschiebt. Dabei verschiebt sich die ursprüngliche Stellkennlinie insgesamt zu kleineren Störpegelwerten, wenn der Lautstärkesteller 12 häufiger in Richtung Lautstärkeerhöhung als in Richtung Lautstärkereduzierung verstellt wird und umgekehrt zu größeren Pegelwerten, wenn der Lautstärkesteller 12 häufiger zur Lautstärkereduzierung als zur Lautstärkeerhöhung verstellt wird.

Liegt ein Störgeräuschpegel vor, bei welchem die Differenzspannung am Ausgang der Summenschaltung 15 den oberen Schwellwert des Schwellwertdiskriminators 25 übersteigt, so ist das AND-Gatter 30 gesperrt und das AND-Gatter 31 gesetzt. Wird in diesem Kennlinienabschnitt c nunmehr der Lautstärkesteller 12 betätigt, so gelangt der von diesem ausgegebene Zählimpuls an den Vorwärts-Rückwärts-Zähler 33, dessen Zählinhalt entsprechend der Verstellrichtung des Lautstärkestellers 12 erhöht oder erniedrigt wird. Der Zählinhalt des Zählers 33 steht als Steuergröße C am Funktionsnetzwerk 18 an und bewirkt eine derartige Einstellung des Funktionsnetzwerks, daß die Stellkennlinie im Kennlinienabschnitt c in Richtung des Pfeils C modifiziert wird. Eine Verschiebung der Stellkennlinie zu größerem Nutzpegel erfolgt dann, wenn der Lautstärkesteller häufiger in Richtung Lautstärkeerhöhung als in Richtung Lautstärkeredu-

zierung verstellt wird. Umgekehrt erfolgt eine Verkleinerung des max. Spannungspegels wenn der Lautstärkesteller 12 eine Verstellung häufiger zur Lautstärkereduzierung als zur Lautstärkeerhöhung erfährt.

Liegt der momentane Störgeräuschpegel im Kennlinienabschnitt a, so wird der untere Schwellwert des Fensterdiskriminators 25 nicht erreicht. Eine Betätigung des Lautstärkestellers 12 bewirkt dann lediglich eine Anhebung oder Absenkung der Grundlautstärke in Richtung A.

Nach wenigen Einstellkorrekturen am Lautstärkesteller 12 hat sich dann eine Stellkennlinie eingestellt, die optimal an die räumlichen Gegebenheiten des Fahrzeuginnenraums und an das Hörempfinden der Kraftfahrzeuginsassen angepaßt ist. Bei Veränderung des Störgeräuschpegels wird dann der Nutzsignalpegel entsprechend der sich letztendlich eingestellten modifizierten Stellkennlinie verändert, wobei dann weitere Korrekturen am Lautstärkeeinsteller 12 zu keinem Zeitpunkt mehr erforderlich sind.

Die Erfindung ist nicht auf das vorstehend beschriebene Ausführungsbeispiel beschränkt. So können die Stellgrößen B und C noch zusätzlich von der Lautstärkestellung und/oder der Lautheit abhängig sein. Auch ist es möglich, die Steuergrößen B und C von der Größe der jeweils vorgenommenen Verstellung des Lautstärkestellers abhängig zu machen . Hierbei braucht lediglich der Lautstärkesteller eine dem Verstellweg entsprechende Anzahl von Zählimpulse auszugeben, die dem Kennlinienadapter zugeführt werden.

Des weiteren ist es nicht erforderlich, daß die Schaltungsanordnung einen Spannungsverstärker mit im Sinne

der Abstimmung der Schaltungsanordnung geregelter Spannungsverstärkung aufweist. Der Spannungsverstärker kann in gleicher Weise wie in der DE-OS 29 42 331 beschrieben ausgebildet sein, wobei dann eine Abstimmung der Schaltungsanordnung nach Einbau im Kraftfahrzeug vor Ort durchgeführt werden muß.

-.-.-.-.-.-.-

Nummer: 33 22 055
Int. Cl.³: H 03 G 3/20
Anmeldetag: 18. Juni 1983
Offenlegungstag: 20. Dezember 1984

R.Nr. 1780

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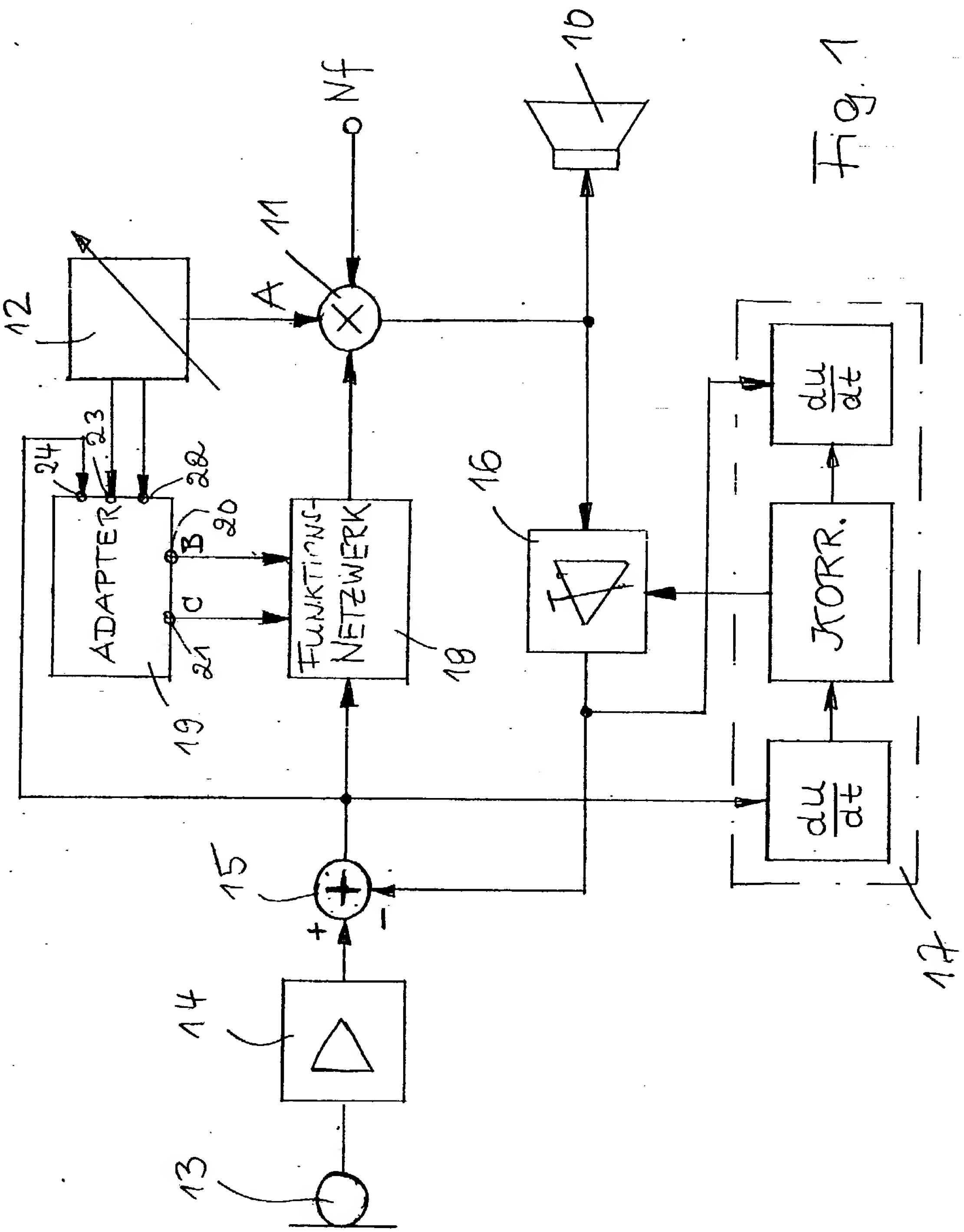

Fig. 1

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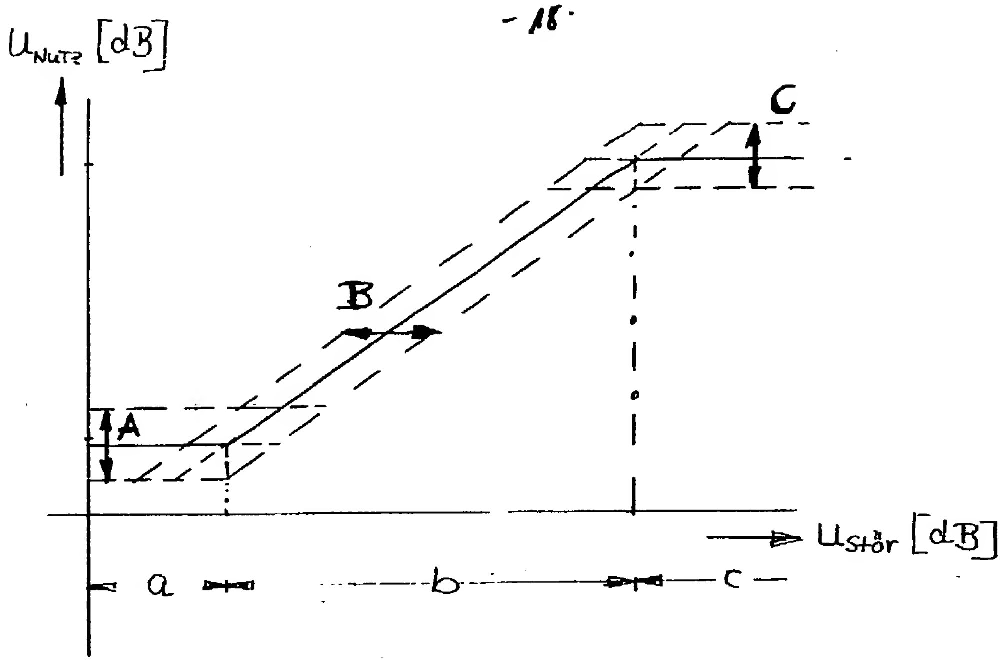

Fig. 2

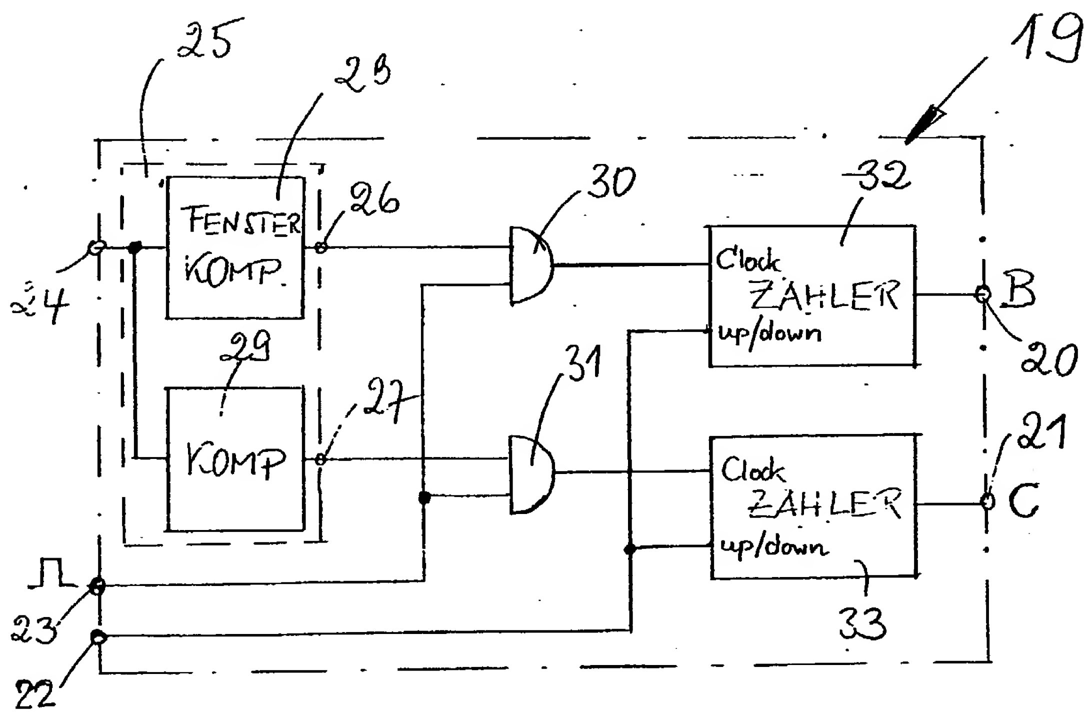

Fig. 3

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Total (1)	14	-	-

United States Patent [19]
Takizawa
[11] 4,254,303
[45] Mar. 3, 1981

[54] **AUTOMATIC VOLUME ADJUSTING APPARATUS**

[75] Inventor: Tetsuya Takizawa, Yamato, Japan
[73] Assignee: Viva Co., Ltd., Noda, Japan

[21] Appl. No.: 64,384
[22] Filed: Aug. 8, 1979
[30] Foreign Application Priority Data

Aug. 26, 1978	[JP]	Japan	53-103385
Nov. 25, 1978	[JP]	Japan	53-144943
Dec. 20, 1978	[JP]	Japan	53-156239

[51] Int. Cl.^{3} H03G 1/02
[52] U.S. Cl. .... 179/1 VL; 179/1 P
[58] Field of Search .... 179/1 VL, 1 P, 1 MN; 330/144, 86, 51, 110

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Primary Examiner—Daryl W. Cook
Attorney, Agent, or Firm—Frishauf, Holtz, Goodman & Woodward

[57] **ABSTRACT**

An automatic volume adjusting apparatus comprises a volume adjusting device connected between a sound signal source and a loudspeaker; a first rectifier circuit for generating a first signal having a D.C. level corresponding to the level of a sound signal supplied from the volume adjusting device to the loudspeaker; a microphone and a second rectifier circuit for generating a second signal having a D.C. level corresponding to the level of a composite sound constituted by the sound generated by the loudspeaker and the ambient noise; a subtraction circuit for generating a third signal having a level corresponding to the level difference between the first and second signals, i.e. the level of the ambient noise; a gate circuit for sampling the third signal at predetermined intervals; and an integrator circuit for holding the output of the gate circuit for a predetermined period of time. The output of the integrator circuit controls the volume adjusting device.

5 Claims, 7 Drawing Figures

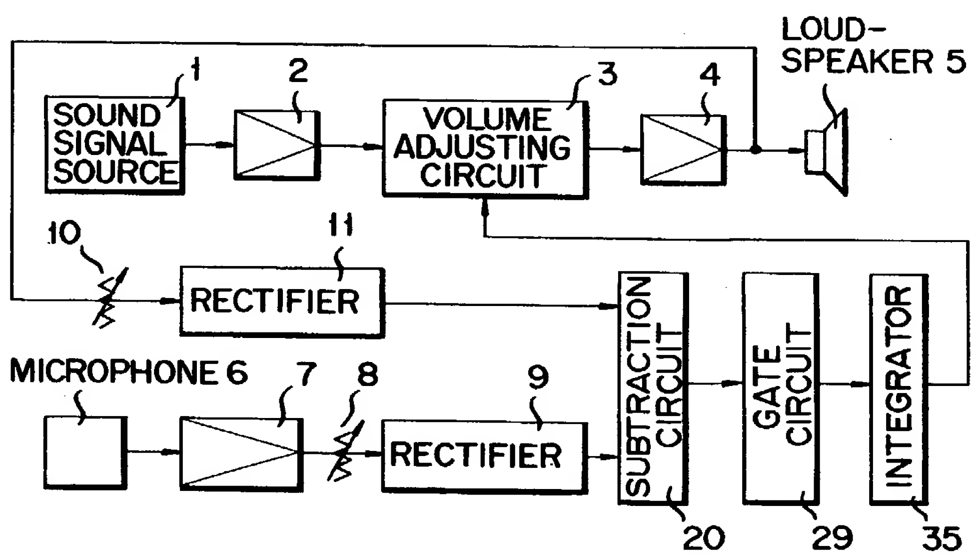


FIG. 1

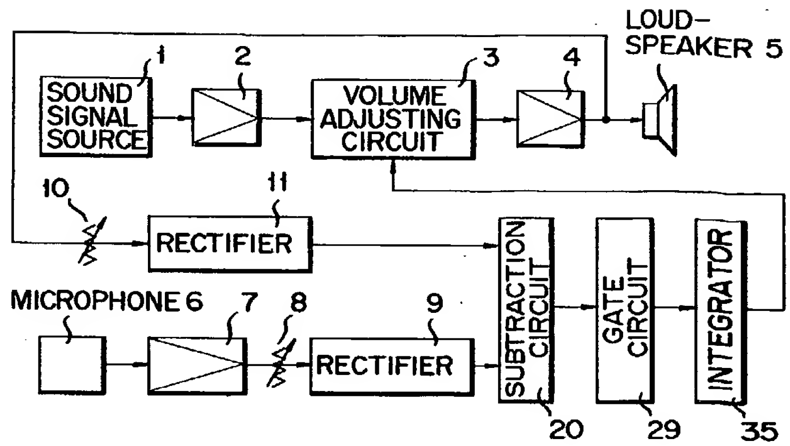
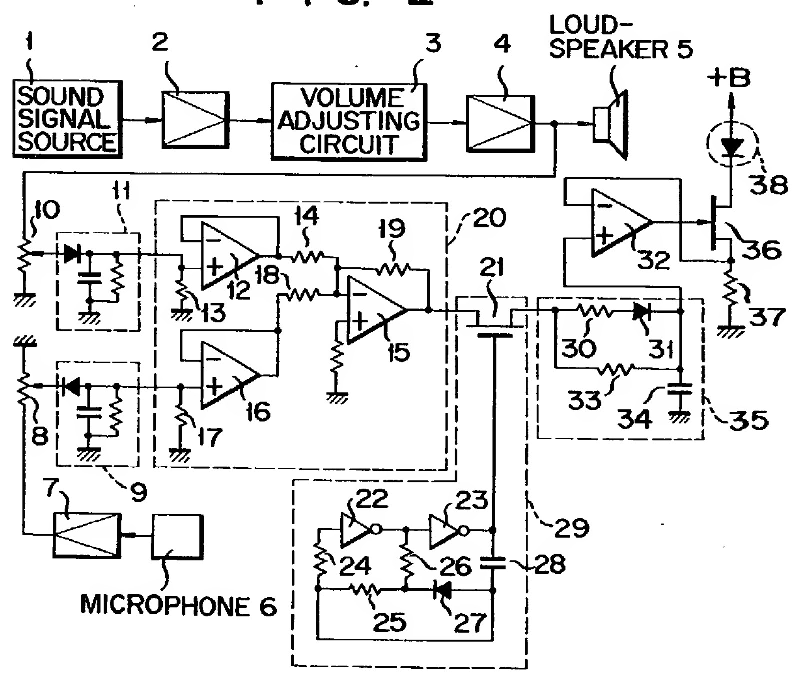

LOUD-SPEAKER 5
1
2
3
4
SOUND SIGNAL SOURCE
VOLUME ADJUSTING CIRCUIT
10
11
RECTIFIER
MICROPHONE 6
7
8
9
RECTIFIER
SUBTRACTION CIRCUIT
GATE CIRCUIT
INTEGRATOR
20
29
35

FIG. 2


1
2
3
4
LOUD-SPEAKER 5
SOUND SIGNAL SOURCE
VOLUME ADJUSTING CIRCUIT
+B
38
10
11
14
19
20
21
32
36
37
12
13
18
15
30
31
33
34
35
16
17
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9
7
22
23
29
24
26
28
25
27
MICROPHONE 6

FIG. 3

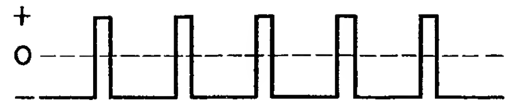


FIG. 4

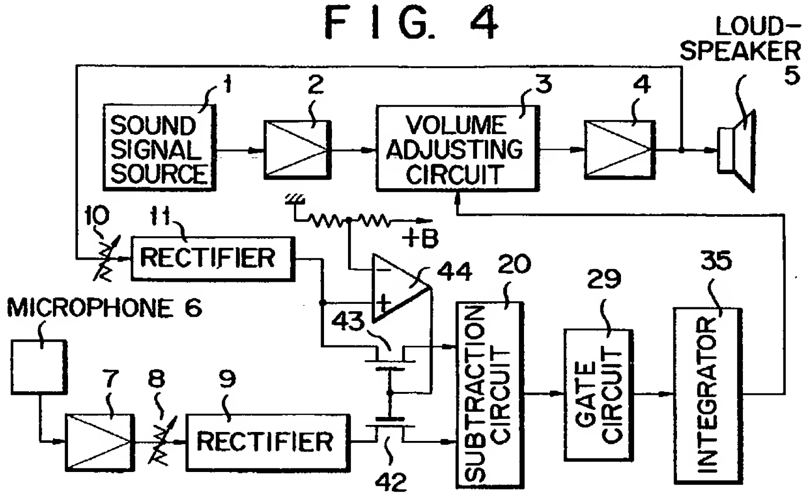


FIG. 5

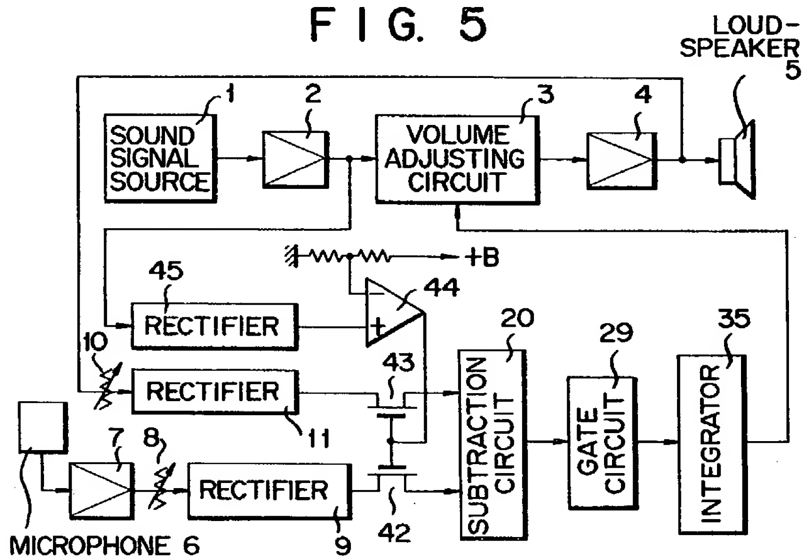


FIG. 6

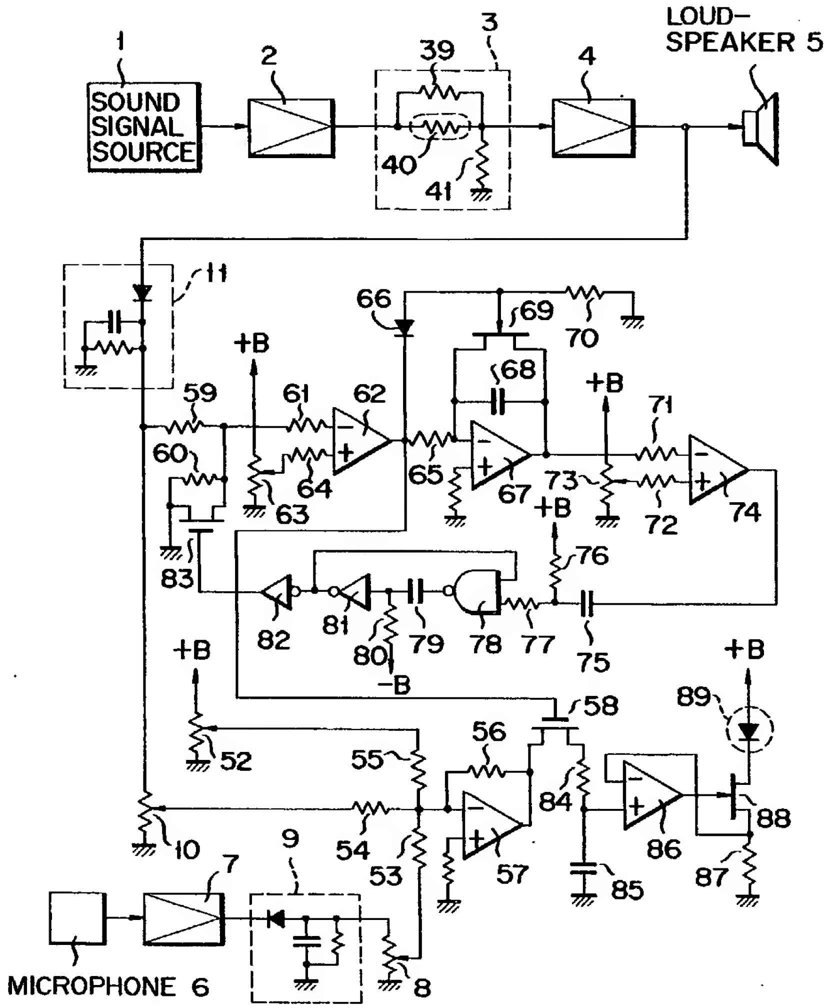
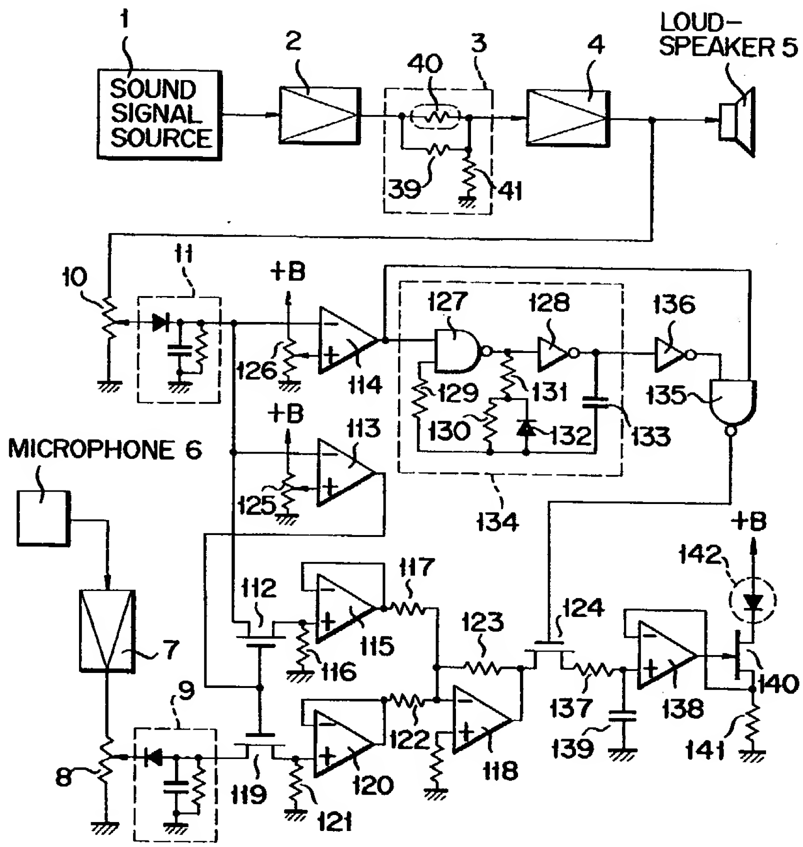


FIG. 7



1
SOUND SIGNAL SOURCE
2
40
3
4
LOUD-SPEAKER 5
39
41
10
11
+B
127
128
136
126
114
129
131
135
+B
113
130
132
133
MICROPHONE 6
125
134
+B
142
117
112
124
123
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121

AUTOMATIC VOLUME ADJUSTING APPARATUS

BACKGROUND OF THE INVENTION

This invention relates to an automatic volume adjusting apparatus which can correctly adjust the volume of sound from a loudspeaker in accordance with the variation of ambient noise level.

Many attempts have been made to automatically adjust the volume of sound from the audio output device of a television set, a radio, a tape recorder, a loudspeaker and the like, in accordance with the variation of ambient noise level. And many methods have been proposed to achieve such an automatic volume adjusting. Notable among them are as follows:

(a) A microphone is used to detect a composite sound constituted by the ambient noise and the sound from a loudspeaker. Only the ambient noise component of the composite sound is removed and detected from a composite sound signal generated by a sound signal circuit. The ambient noise component is used to control the gain of an audio output device.

(b) The ambient noise is detected when a loudspeaker does not generate sound. According to the level of the ambient noise one of static circuits such as switches which have a self-holding function is selected thereby to control the volume of sound generated by the loudspeaker.

With the method (a), however, it is difficult to remove and detect only the ambient noise component from the composite sound constituted by the ambient noise and the sound from the loudspeaker, that is, to completely cancel out the electric signal from the sound signal circuit and the sound generated by the loudspeaker. This is because the sound from the loudspeaker involves in phase delay, the loudspeaker and microphone differ in frequency characteristic, or for some other reasons. If the electric signal and the sound could be cancelled out almost completely, the ambient noise would be masked by the sound from the loudspeaker when the sound volume increases to a certain level. As a result, the microphone would fail to detect the ambient sound, and the sound volume could no longer increase. Consequently, the volume adjustment would become unstable.

In the method (b) the volume adjustment is carried out somewhat statically. The method (b) is not therefore practical in case the ambient noise changes abruptly or periodically while a loudspeaker is generating sound. With the method (b) it is next to impossible to adjust the sound volume particularly when the loudspeaker generates a continuous sound such as music. This is because in this case the loudspeaker is rarely silent so that the volume of sound from it is maintained at constant level.

Accordingly, it is an object of this invention to provide an automatic volume adjusting apparatus which is so designed as to increase or reduce the volume of sound generated by a loudspeaker accurately in accordance with the level of the ambient noise, regardless of masking phenomenon which occurs when the volume of sound from the loudspeaker increases to a certain level according to the ambient noise level.

SUMMARY OF THE INVENTION

According to this invention, there is provided an automatic volume adjusting apparatus which comprises a volume adjusting device connected between a sound signal source and a loudspeaker, means for generating a first signal corresponding to the level of a sound signal supplied from the sound signal source to the loudspeaker, means for generating a second signal corresponding to the level of a composite sound constituted by the sound generated by the loudspeaker and the ambient noise, means for processing the first and second signals so as to generate a third signal corresponding substantially to only the level of the ambient noise, means for sampling the third signal at predetermined intervals, means for holding the output of the sampling means for a predetermined period of time, and means for controlling, according to the output of the holding means, the level of the sound signal passing through the volume adjusting device.

With the apparatus of such construction, the volume adjusting device is controlled by the third signal which is sampled and then held immediately before a maskings phenomenon takes place. Thus, the volume of sound from the loudspeaker never fails to be controlled according to the ambient noise level in spite of the masking phenomenon which occurs when the volume of sound from the loudspeaker is extremely large.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a block circuit diagram of an automatic volume adjusting apparatus according to this invention;

FIG. 2 is a circuit diagram showing the apparatus of FIG. 1 more in detail;

FIG. 3 illustrates the output waveform of a non-stable multivibrator shown in FIG. 2;

FIGS. 4 and 5 are block circuit diagrams showing other embodiments of this invention; and

FIGS. 6 and 7 are circuit diagrams of still other embodiments of this invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

As shown in FIGS. 1 and 2, a sound signal is supplied from a sound signal source 1 such as a television set, a radio, a tape recorder and a microphone through a preamplifier 2 to a volume adjusting circuit 3. The level of the sound signal is adjusted by the circuit 3, and the sound signal is then amplified by a power amplifier 4 and finally converted into sound by a loudspeaker 5. The sound thus generated by the loudspeaker 5 and the ambient noise are detected by a microphone 6, which produces a detection signal corresponding to a composite sound constituted by the sound from the loudspeaker 5 and the ambient noise. The detection signal is amplified by a microphone amplifier 7 and supplied through a level adjustor 8 to a rectifier 9. The rectifier 9 converts of the detection signal into, for example, a negative D.C. output. The sound signal from the power amplifier 4 is supplied through a level adjustor 10 to a rectifier 11. The rectifier 11 converts the sound signal into, for example, a positive D.C. output. The output of the rectifier 11 is coupled to the non-inversion input of a buffer 12 and to the ground via a resistor 13. The output of the buffer 12 is connected to the inversion input of the buffer 12 and to the non-inversion input of an inversion adder-amplifier 15 through a resistor 14. The output of the rectifier 9 is coupled to the non-inversion input of a buffer 16 and to the ground through a resistor 17. The output of the buffer 16 is connected to the inversion input of the buffer 16 and to the inversion input of the

inversion adder-amplifier 15 via a resistor 18. The output of the inversion adder-amplifier 15 is connected to the non-inversion input of the adder-amplifier 15 via a resistor 19 and to the drain of an analog switch 21. The buffers 12, 16, inverson adder-amplifier 15, and resistors 13, 14, 17, 18, 19 constitute a subtraction circuit 20.

As shown in FIG. 2, inverters 22 and 23, resistors 24, 25 and 26, a diode 27 and a capacitor 28 constitute a non-stable multivibrator of the known type. The output of the multivibrator is coupled to the gate of the analog switch 21. The non-stable multivibrator and the analog switch 21 constitute a gate circuit 29. The source of the analog switch 21 is connected to the non-inversion input of a buffer 32 through a resistor 30 and a diode 31 and also to the non-inversion input of the buffer 32 through a resistor 33. The non-inversion input of the buffer 32 is connected to the ground via a capacitor 34. The diode 31, resistors 30 and 33 and the capacitor 34 constitute an integrator 35 having a time constant which is determined by the rated values of the constituent elements. The invention input of the buffer 32 is coupled to the source of a field effect transistor 26 (hereinafter called "FET") and to the ground through a resistor 37. The output of the buffer 32 is connected to the gate of the FET 36. The drain of the FET 36 is connected to the cathode of a light-emitting diode 38, the anode of which is connected to a positive D.C. power source +B. The light-emitting diode 38 is integrally formed with a light-receiving element 40 such as a CdS element. The light-receiving element 40 and two resistors 39 and 41 constitute the volume adjusting circuit 3.

The buffers 12, 16 and 32, inversion adder-amplifier 15, inverters 22 and 23 and analog switch 21 are supplied with power from the positive D.C. power source +B and a negative D.C. power source −B.

Now it will be described how the embodiment of FIGS. 1 and 2 operates. According to the frequency characteristics of the loudspeaker 5 and the microphone 6 and the positional relationship between the loudspeaker 5 and the microphone 6, the level adjustors 8 and 10 and the rectifiers 9 and 11 are controlled so that the absolute values of D.C. outputs of the rectifiers 9 and 11 become substantially equal when there is no ambient noise.

The D.C. output of the rectifier 9 is supplied to the buffer 16, and the D.C. output of the rectifier 11 to the buffer 12. At the output terminals of the buffers 12 and 16 there are obtained outputs of the same polarity. The inversion adder-amplifier 15 obtains an output which is determined by the sum of the outputs of the buffers 12 and 16, the resistances of the resistors 14, 18 and 19 and the amplification factor of the adder-amplifier 15. This output is the output of the subtraction circuit 20. Since the negative D.C. output of the rectifier 9 and the positive D.C. output of the rectifier 11 have substantially the same absolute value when no ambient noise exists, the sum of them becomes a negative voltage when there is an ambient noise. Thus, the output of the inversion adder-amplifier 15, i.e. the output of the subtraction circuit 20, is a positive voltage which is proportional to the ambient noise component.

The multivibrator of the gate circuit 29 produces a pulse output having such a waveform as illustrated in FIG. 3. As shown in FIG. 3, the pulse output has a relatively short positive potential period and a long negative potential period. This pulse output is supplied to the gate of the analog switch 21. During the positive potential period of the pulse output the drain-source path of the analog switch 21 is conductive. During the negative potential period of the pulse output the drain-source path of the switch 21 is non-conductive. As a result, the gate circuit 29 effects the sampling of the output of the subtraction circuit 20.

The resistor 30 of the integrator 35 has such a resistance that the forward resistance of the diode 31 can be neglected with respect to the resistance of the resistor 30. Further, the resistor 33 of the integrator 35 has a resistance far higher than that of the resistor 30. The charge time constant charged between the terminals of the capacitor 34 is therefore determined substantially by the resistor 30 and the capacitor 34. On the other hand, the discharge time constant of the integrator 35 is determined substantially by the resistor 33 and the capacitor 34 since the backward resistance of the diode 31 is extremely high. Consequently, the charge time constant becomes much smaller than the discharge time constant. In other words, the write-in time constant of the integrator 35 is small, and the read-out time constant is large. The relationship between the pulse width during the positive potential period and the charge, discharge time constant is determined by the ratio between the positive pulse width and the charge time constant. The ratio between the positive pulse width and the charge time constant is properly selected so that the latter is sufficiently large with respect to the former.

The positive D.C. output of the subtraction circuit 20 is charged between the terminals of the capactior 34 while the analog switch 21 is closed, according to the time constant determined by the resistor 30 and the capacitor 34. While the analog switch 21 is not conductive, the drain-source impedance of the switch 21 and the input impedance of the buffer 32 are extremely high. The potential of the capacitor 34 is therefore maintained at the value immediately before the analog switch 21 has become non-conductive.

In this way, the positive D.C. output of the subtraction circuit 20 is charged in the capacitor 34 according to the time constant determined by the resistor 30 and the capacitor 34, every time the analog switch 21 becomes conductive. Since the charge time constant for charging the output of the subtraction circuit 20 to the capacitor 34 is small, the potential across the terminals of the capacitor 34 elevates in a quick response to the increase of ambient noise.

The positive D.C. voltage obtained across the terminals of the capacitor 34 is supplied to the non-inversion input of the buffer 32. Since the buffer 32, FET 36 and resistor 37 constitute a constant current circuit which uses the light-emitting diode 38 as its load, there flows in the light-emitting diode 38 a current which is proportional to the positive D.C. voltage suppled to the buffer 32 and the resistance of the resistor 37. The resistance of the CdS element 40, which is integrally formed with the light-emitting diode 38, becomes lower as the current flowing in the light-emitting diode 38 increases, and the attenuation ratio of the volume adjusting circuit 3 becomes smaller. As a result, the input level of the power amplifier 4 elevates, thus increasing the sound volume of the loudspeaker 5.

The louder the sound from the loudspeaker 5 grows, the higher become the negative D.C. output level of the rectifier 9 and the positive D.C. level of the rectifier 11. If there is no ambient noise, the sound component in the output of the subtraction circuit 20 is substantially zero since the absolute values of the D.C. outputs of the rectifiers 9 and 11 are identical. If an ambient noise

exists, the output of the subtraction circuit 20 contains a noise component corresponding to the ambient noise. Thus, the loudspeaker 5 generates sound the volume of which is proportional to the ambient noise component in the output of the subtraction circuit 20. When the ambient noise ceases under this condition, the output level of the subtraction circuit 20 is reduced to substantially zero. As a result, the capacitor 34 is discharged through the resistor 33 while the analog switch 21 remains conductive. Since the discharge time constant is large, the volume of sound from the loudspeaker 5 gradually increases until it reaches the initial value.

When the ambient noise becomes very large and so does the volume of sound from the loudspeaker 5, the sound from the loudspeaker 5 masks the ambient noise, thus making it difficult for the microphone 6 to detect the ambient noise. This phenomenon is equivalent to the level drop of ambient noise. As a result, the volume of sound from the loudspeaker 5 will likely be reduced. In effect, however, the potential between the terminal of the capacitor 34 would never be lowered abruptly. This is because the discharge time sonstant determined by the resistor 33 and the capacitor 34 is much larger than the value corresponding to the period during which the analog switch 21 is conductive, and the period during which the analog switch 21 is non-conductive is much larger than the period during which the switch 21 remains conductive.

Further, the charge time constant determined by the resistor 30 and the capacitor 34 is selected to have a proper value with respect to the period during which the analog switch 21 remains conductive. In addition, it is quite rare that the sound signal source 1 keeps generating high level sound signals for a long time even if the signals represent a continuous musical sound. The sound signal source 1 produces signals of a low level for a period of time, though a short one. During such a period of time the ambient noise component in the output of the subtraction circuit increases equivalently. Thus, only if analog switch 21 is conductive during this period, the capacitor 34 is charged again with the ambient noise component in a very short time to have its potential raised. Consequently, the masking of the ambient noise can be suppressed even when the volume of sound from the loudspeaker 5 is relatively large, whereby the volume of sound is adjusted in a natural way.

Needless to say, the volume adjusting circuit 3 may be constituted by transistors which are voltage-controlled to vary the level of an input sound signal, instead of a photo-coupler constituted by an electric-to-light converter such as a light-emitting diode and a light-to-electric converter such as a CdS element.

FIGS. 4 and 5 illustrate other embodiments of this invention. Unlike the embodiment shown in FIGS. 1 and 2, these embodiments can prevent the volume of sound from the loudspeaker from momentarily growing too large when the loudspeaker starts generating sound. Now referring to FIGS. 4 and 5 wherein like or the same numerals are used to denote like or the same parts as those of the embodiment shown in FIGS. 1 and 2, these embodiments will be described.

In the embodiment of FIG. 4, an analog switch 42 is connected between a rectifier 9 and a subtraction circuit 20, and another analog switch 43 between the subtraction circuit 20 and a rectifier 11. Further, a comparator 44 is connected to the output of the rectifier 11. When the output level of the rectifier 11 is substantially zero, the output of the comparator 44 renders both analog switches 42 and 43 non-conductive, whereby the outputs of the rectifiers 9 and 11 are not supplied to the subtraction circuit 20. In the embodiment of FIG. 5, a rectifier 45 is provided to rectify the input sound signal to a volume adjusting circuit 3. The output of the rectifier 45 operates a comparator 44. When the output level of the rectifier 45 is zero, analog switches 42 and 43 rendered non-conductive, whereby the outputs of rectifiers 9 and 11 are not supplied to a subtraction circuit 20. In all the other respects the embodiments of FIGS. 4 and 5 are identical with the embodiment of FIGS. 1 and 2.

In the embodiments of FIGS. 1 and 2, FIG. 4 and FIG. 5, the gate circuit 29 samples out the output of the subtraction circuit 20 which corresponds to an ambient noise component. The output of the gate circuit 29 is held by the integrator 35 and controls the volume adjusting circuit 3 even when the ambient noise is masked by the sound from the loudspeaker 5, thus preventing a level drop of the sound from the loudspeaker 5. But, since the masking phenomenon occurs when the output sound level of the loudspeaker 5 rises beyond a specific value, it should be better to use the output of the subtraction circuit 20 to control the volume adjusting circuit 3 so long as the output sound level of the loudspeaker 5 is below the specific value.

This invention provides an automatic volume adjusting apparatus of another type in which the volume adjusting mode is changed to another in accordance with the output sound level of a loudspeaker. FIG. 6 shows such an automatic volume adjusting apparatus.

As shown in FIG. 6, the output of a rectifier 9 is supplied to a level adjustor 8, and its level is adjusted by the level adjustor 8. The output of the level adjustor 8 is supplied to the inversion input of an inversion adder-amplifier through a resistor 53. The output of a rectifier 11 is supplied to a level adjustor 10, which adjusts the level of the output of the rectifier 11. The output of the level adjustor 10 is supplied to the inversion input of the inversion adder-amplifier 57. Another level adjustor 52 is connected between the ground and a positive D.C. power source +B. The output positive D.C. voltage of the level adjustor 52 is supplied via a resistor 55 to the inversion input of the inversion adder-amplifier 57. The output of the adder-amplifier 57 is coupled to the inversion input of the adder-amplifier 57 through a resistor 56 and to the drain of an analog switch 58. The output level of the adder-amplifier 57 is determined by the sum of the outputs of the level adjustors 8, 10 and 52, the resistance of the resistors 53–56 and the amplification factor of the adder-amplifier 57.

The output of the rectifier 11 is voltage-divided by resistors 50 and 51 and then supplied via a resistor 61 to the inversion input of a level comparator 62. A voltage adjustor 63 is connected between the ground and the positive D.C. power source +B to provide a D.C. voltage. The D.C. voltage is applied to the non-inversion input of the level comparator 62 through a resistor 64. The output of the level comparator 62 is coupled to the gate of the analog switch 58, to the inversion input of an integrator 67 via a resistor 65 and to the cathode of a diode 66. Between the inversion input and output of the integrator 67 there is connected a capacitor 68. Further, a FET 69 has its source and drain connected to the inversion input and output of the integrator 67, respectively. The gate of the FET 69 is connected to the anode of the diode 66 and to the ground via a resistor

70. The output of the integrator 67 is coupled to the inversion input of a level comparator 74 via a resistor 71. The non-inversion input of the level comparator 74 is connected to receive via a resistor 72 a positive D.C. voltage from a voltage adjustor 73 which is connected between the ground and the positive D.C. power source +B. The output of the level comparator 74 is coupled to one terminal of a capacitor 75. The terminal of the capacitor 75 is connected to one input of a two-input NAND circuit 78 through a resistor 77 and to the positive D.C. power source +B through a resistor 76. The output of the NAND circuit 78 is connected to the input of an inverter 81 via a capacitor 79. The input of the inverter 81 is connected to a negative D.C. power source −B through a resistor 80. The output of the inverter 81 is connected to the other input of the NAND circuit 78 and to the input of an inverter 82. The output of the inverter 82 is connected to the gate of an analog switch 83, the drain and source of which are connected to the both ends of a resistor 60, respectively.

The source of the analog switch 58 is connected via a resistor 84 to the non-inversion input of a non-inversion amplifier 86. The non-inversion input of the amplifier 86 is connected to the ground through a capacitor 85. The inversion input of the non-inversion amplifier 86 is connected to the source of a FET 88 and to the ground through a resistor 87. The output of the amplifier 86 is coupled to the gate of the FET 88. The drain of the FET 88 is connected to the cathode of a light-emitting diode 89, the anode of which is connected to the positive D.C. power source +B. As in the embodiment of FIG. 2, the light-emitting diode 89 is coupled optically to a light-receiving CdS element 39.

Now it will be described how the automatic volume adjusting apparatus of FIG. 6 operates.

According to the frequency characteristics of the loudspeaker 5 and the microphone 6 and the positional relationship between the loudspeaker 5 and the microphone 6, the level adjustors 8 and 10 are controlled so that the D.C. output of the rectifier 11 has substantially the same absolute value as, or an absolute value larger than, that of the rectifier 9 with respect to frequencies within the audible frequency range. If the level adjustor 52 produces a positive D.C. voltage having substantially the same absolute value as that of a negative D.C. output of the rectifier 9 which is produced when the loudspeaker 5 generates no sound and which corresponds to the volume of ambient noise, the sum of the outputs of the level adjustors 8, 10 and 52 will be substantially zero or a positive value. If the voltage of the voltage adjustor 63 is higher than the maximum value of a positive D.C. voltage obtained by dividing the output voltage of the rectifier 11 by means of resistors 59 and 60, the output of the level comparator 62 is a positive potential. Then, the gate of the analog switch 58 has a positive potential, whereby the drain-source path of the analog switch 58 becomes conductive. As a result, the non-inversion amplifier 86, the FET 88 and the resistor 87 constitute a constant current circuit, and there flows in the light-emitting diode 89 a current proportional to the resistance of the resistor 87 and the positive D.C. voltage applied to the non-inversion input of the non-inversion amplifier 86. The resistor 84 and the capacitor 85 both connected to the non-inversion input of the non-inversion amplifier 86 constitute a time constant circuit for smoothening the varying output of the adder-amplifier 57.

Since the output of the adder-amplifier 57 is substantially zero or has a negative value, the input voltage of the non-inversion amplifier 86 becomes substantially 0V or lower than 0V. As a result, no current flows through the light-emitting diode 89. The CdS element 40 therefore underdoes no resistance change. Consequently, the attenuation ratio of the volume adjusting circuit 3 remains unchanged.

When there is an ambient noise of a relatively low level, the microphone 6 catches a composite sound consisting of an ambient noise component and the sound from the loudspeaker 5. The negative D.C. output of the level adjustor 8 grows by the value which corresponds to the ambient noise component. The input voltage of the adder-amplifier 57 therefore becomes a negative voltage. Since the adder-amplifier 57 is an inversion amplifier, its output is the sum of a negative D.C. voltage corresponding to the ambient noise component and a positive voltage determined by the amplification factor of the adder-amplifier 57 and the resistance of the resistors 53 to 56. If the analog switch 58 is conductive, the positive output voltage of the adder-amplifier 57 is applied through the analog switch 58, smoothened by the resistor 84 and the capacitor 85, and applied to the inoput of the non-inversion amplifier 86. Then, the D.C. current determined by the input voltage of the amplifier 86 and the resistance of the resistor 87 flows through the light-emitting diode 89, and light corresponding to the D.C. current is emitted from the diode 89 to the CdS element 40. The resistance of the CdS element 40 is lowered to reduce the attenuation ratio of the volume adjusting circuit 3. The input level of the power amplifier 4 therefore rises, thereby increasing the volume of sound from the loudspeaker 5.

The louder the sound from the loudspeaker 5 grows, the higher is the output level of the microphone 6. Thus, the negative D.C. output level of the level adjustor 8 rises by the value corresponding to the increase of sound volume of the loudspeaker 5. At the same time the positive D.C. output level of the level adjustor 10 also rises by the value corresponding to the increase of sound volume. As a result, the sound component in the output of the adder-amplifier 57 is reduced substantially to zero. Thus, the loudspeaker 5 generates sound the volume of which is proportional to the ambient noise component. When the ambient noise ceases under this condition, the output level of the adder-amplifier 57 is reduced substantially to zero. If the analog switch 58 is conductive, the capacitor 85 is discharged through the resistor 84 and comes to have zero potential, thereby reducing the volume of sound from the loudspeaker 5.

When the microphone 6 picks up an ambient noise of a high level, the volume of the loudspeaker 5 is increased very much to raise the output level of the rectifier 11, whereby the inversion input level of the level comparator 62 becomes higher than the non-inversion input level of the level comparator 62. If this happens, the output level of the level comparator 62 changes from a positive potential to a negative one. The gate of the analog switch 58 comes to have a negative potential, and the analog switch 58 is rendered non-conductive. The capacitor 85 therefore holds a voltage corresponding to the output voltage of the adder-amplifier 57 immediately before the analog switch 58 has become non-conductive, and the volume of the loudspeaker 5 remains at a high level which corresponds to the voltage held by the capacitor 85.

Suppose the analog switch 58 is not provide and that the output of the adder-amplifier 57 is coupled directly to the resistor 84. Then, the microphone 6 fails to detect the ambient noise if the sound from the loudspeaker 5 is larger enough to mask the ambient noise. In this case, the automatic volume adjusting apparatus fails to respond to the ambient noise as if there were no ambient noise. The volume of the loudspeaker 5 is reduced to such an extent that the ambient noise is not completely masked, and the microphone 6 detects the ambient signal thereby to increase the volume of the loudspeaker 5 again. The sound volume reduction and sound volume increase are repeated, thus annoying the listeners very much. To avoid such a repetition of volume changes, the analog switch 58 is provided.

If the analog switch 58 were rendered non-conductive by a large volume of the loudspeaker 5 and remained non-conductive, the output of the adder-amplifier 57 would not be supplied to the non-inversion amplifier 86 even if the ambient noise level lowers to drop the output voltage of the adder-amplifier 57. The volume of the loudspeaker 5 would not therefore be reduced. The elements provided at the stages succeeding the level comparator 62 are to prevent such an undersirable phenomenon. It will be now described how these elements function.

When the inversion input voltage of the level comparator 62 becomes higher than the non-inversion input voltage, the output level of the level comparator 62 changes from a positive value to a negative value. At the same time, the integrator 67, which has a time constant determined by the resistor 65 and the capacitor 68, starts operating, and its output level changes from a negative value to a positive value. When the output voltage of the integrator 67 rises over the positive voltage of the level comparator 74, the output level of the level comparator 74 changes from a positive value to a negative one. Then, the capacitor 75 and the resistor 76 and 77 generate a trigger pulse, which is supplied to the input of the NAND circuit 78. The NAND circuit 78, inverters 81 and 82, resistor 80 and capacitor 79 constitute a monostable multivibrator. Upon receipt of the trigger pulse, the monostable multivibrator produces a positive pulse having a width which is determined by the resistor 80 and the capacitor 79. The positive pulse is supplied from the inverter 82 to the gate of the analog switch 83. As long as the positive pulse lasts, the analog switch 83 remains conductive, thus short-circuiting the resistor 60. The resistor 59 has such a resistance that the output of the rectifier 11 is not affected even if the resistor 60 is short-circuited. Once the resistor 60 has been short-circuited, the inversion input potential of the level comparator 62 is reduced substantially to zero. The analog switch 58 is therefore rendered conductive and remains so so long as said positive pulse lasts. When the output level of the level comparator 62 changes from a negative value to a positive value, the diode 66 is biased backwardly and made non-conductive. The gate voltage of the FET 69 therefore elevates to zero from a negative value, and the drain-source path of the FET 69, which has been non-conductive, is rendered conductive. As a result, the capacitor 68 is discharged through the drain-source path of the FET 69. The output of the integrator 67 comes to have a negative value, and the integrator 67 is therefore reset.

The above-mentioned series of operations are repeated every time the inversion input voltage of the level comparator 62 becomes higher than the non-inversion input voltage. The width of the positive pulse is sufficiently smaller with respect to the period of time during which said series of operations are repeated once. Every time the positive pulse is generated, the analog switch 58 is rendered conductive. This means that the analog switch 58 detected the output level of the adder-amplifier 57 repeatedly for a short time at regular intervals. The constant current circuit is controlled by the output of the adder-amplifier 57 while the analog switch 58 remains conductive. Since the width of the pulse is sufficiently small in comparison with the time constant determined by the resistor 84 and the capacitor 85, the potential across the terminals of the capacitor 85 will never change abruptly. Thus, the current flowing through the light-emitting diode 89, i.e. load of the constant current circuit, will never change abruptly, either. The output level of the volume adjusting circuit 3 therefore changes slowly, thereby suppressing the masking of the ambient noise. Consequently, the volume of the loudspeaker 5 can be automatically adjusted in a natural manner.

As described above, in the embodiment of FIG. 6, when the level comparator detects a high level output of the volume adjusting circuit; the level comparator holds the output of the subtraction circuit, and at the same time various operations are repeated for a predetermined time thereby to deliver the output of the subtraction circuit from the level comparator, whereby the volume of the loudspeaker is adjusted by the output of the subtraction circuit. The apparatus of FIG. 6 can therefore avoid an unnatural variation of sound volume even if the ambient noise is masked by a large volume of the loudspeaker, and the volume of the loudspeaker can be automatically adjusted in a natural manner.

FIG. 7 illustrates another embodiment of the invention which improves the level comparison carried out in the embodiment of FIG. 6.

As shown in FIG. 7, the output of a rectifier 11 is connected to the drain of an analog switch 112, the non-inversion input of a level comparator 113 and the non-inversion input of a level comparator 114. The source of the analog switch 112 is connected to the non-inversioninput of a buffer 115 and the ground through a resistor 116. The output of the buffer 115 is connected to the inversion input of the buffer 115 and to the inversion input of an inversion adder-amplifier 118 through a resistor 117. The output of a rectifier 9 is connected to the drain of an analog switch 119. The source of the analog switch 119 is connected to the non-inversion input of a buffer 120 and to the ground through a resistor 121. The output of the buffer 120 is connected to the inversion input of the buffer 120 and to the inversion input of the adder-amplifier 118 through a resistor 122. The output of the inversion adder-amplifier 118 is connected to the inversion input of the adder-amplifier 118 via a resistor 123 and to the drain of an analog switch 124. The inversion input of the level comparator 113 is connected to a movable contact of a level adjustor 125 which is connected between a positive D.C. power source +B and the ground. The output of the level comparator 113 is connected to the gates of the analog switches 112 and 119. The inversion input of the level comparator 114 is connected to a movable contact of a level adjustor 126 which is connected between the positive D.C. power source +B and the ground.

The output of the level comparator 114 is connected to the control input of a non-stable multivibrator 134

which is constituted by a two-input NAND circuit 127, an inverter 128, resistors 129, 130 and 131, a diode 132 and a capacitor 133. The output of the level comparator 114 is connected also to one input of a NAND circuit 135. The output of the multivibrator 134 is coupled to the input of an inverter 136, the output of which is connected to the other input of the NAND circuit 135. The output of the NAND circuit 135 is connected to the gate of the analog switch 124. The source of the analog switch 124 is connected to the non-inversion input of a buffer 138 through a resistor 137. The non-inversion input of the buffer 138 is connected to the ground via a capacitor 139. The inversion input of the buffer 138 is connected to the source of a FET 140 and to the ground through a resistor 141. The output of the buffer 138 is connected to the gate of the FET 140. The drain of the FET 140 is connected to the cathode of a light-emitting diode 142, the anode of which is connected to the positive D.C. power source +B. A CdS element 40 for receiving light from the light-emitting diode 142 is integrally formed with the diode 142. The CdS element 40 and resistors 39 and 41 constitute a volume adjusting circuit 3.

Now it will be described how the embodiment of FIG. 7 operates.

First, it will be described how the apparatus operates when there if no ambient noise. Also in this embodiment, the level adjustors 8 and 10 are so designed as to render the D.C. output of the rectifier 11 either substantially equal to, or larger than the D.C. output of the rectifier 9. Further, the level adjustors 125 and 126 render the voltage across the ground and the inversion input of the level comparator 113 lower than the voltage across the ground and the inversion input of the level comparator 114.

When the voltage of the level comparator 113 is lower than the minimum output D.C. voltage of the rectifier 11 and when the voltage of the level comparator 114 is higher than the maximum output D.C. voltage of the rectifier 11, the output of the level comparator 113 has a positive potential. Thus, the drain-source paths of the analog switches 112 and 119 are made conductive, whereby the output of the rectifiers 11 and 9 are supplied to the buffers 115 and 120, respectively. As a result, the buffers 115 and 120 produce outputs of the same polarity. These outputs of the buffers 115 and 120 are supplied to the inversion input of the inversion adder-amplifier 118 through the resistors 117 and 122, respectively. The adder-amplifier 118 produces an output the level of which is determined by the amplification factor of the adder-amplifier 118 and the resistances of the resistors 117, 122 and 123. Since the absolute value of the positive D.C. output of the rectifier 11 is substantially equal to, or larger than, that of the negative D.C. output of the rectifier 9, the sum of the inputs to the adder-amplifier 118 becomes zero or has a positive value. The output of the inversion adder-amplifier therefore becomes zero or has a negative value.

Since the voltage of the level comparator 114 is higher than the D.C. output voltage of the rectifier 11, the output of the level comparator 114 has a negative potential. The non-stable multivibrator 134 therefore does not oscillate, and its output has a negative potential. Thus, the NAND circuit 135 receives the negative potential from the level comparator 114 and a positive potential obtained by inverting the negative potential from the multivibrator 134 through the inverter 136. As a result, the output of the NAND circuit 135 has a positive potential and renders the drain-source path of the analog switch 124 conductive. The output of the inversion adder-amplifier 118 is therefore charged in the capacitor 139 through the drain-source path of the analog switch 124 and through the resistor 137 and is supplied to the non-inversion input of the buffer 138. The buffer 138, FET 140 and resistor 141 constitute a constant current circuit which uses the light-emitting diode 142 as a load. The positive D.C. voltage applied to the non-inversion input of the buffer 138 is to be applied on the light-emitting diode 142, and the current proportional to the resistance of the resistor 141 is to flow through the light-emitting diode 142. In this case, however, no current flows through the light-emitting diode 142. This is because the FET 140 is made non-conductive since the input level of the buffer 138 is zero or of a negative value and thus its output level is also zero or of a negative value. Consequently, the resistance of the CdS element 40 does not change, and the attenuation ratio of the volume adjusting circuit 3 remains unchanged, whereby the output volume of the loudspeaker 5 is maintained.

Second, it will be described how the apparatus of FIG. 7 operates in case there is an ambient noise. The microphone 6 picks up a composite sound consisting of the ambient noise and the sound from the loudspeaker 5. The negative D.C. output of the rectifier 9 becomes larger by the value corresponding to the ambient noise component. The sum of the inputs to the inversion adder-amplifier 118 becomes a negative potential, and the inversion adder-amplifier 118 generates a positive D.C. voltage which is proportional to the ambient noise component. If the analog switch 124 is conductive, the positive D.C. output voltage of the adder-amplifier 118 is supplied to the input of the buffer 138 through the analog switch 124 with a time constant determined by the resistor 137 and the capacitor 139. Then, a current determined by the input voltage to the buffer 138 and the resistance of the resistor 141 flows through the light-emitting diode 142. According to this current, or amount of light emitted from the diode 142, the resistance of the CdS element 40 is reduced, thus decreasing the attenuation ratio of the volume adjusting circuit 3. The input level of the power amplifier 4 therefore rises, and do does the output level of the power amplifier 4. As a result, the output volume of the loudspeaker 5 is increased.

The louder the sound from the loudspeaker 5 grows, the higher the output level of the microphone 6 becomes. The negative D.C. output level of the rectifier 9 rises in proportion of the volume increase. At the same time, the positive D.C. output level of the rectifier 11 rises in proportion to the volume increase, too. The sound component in the output of the adder-amplifier 118 is reduced substantially to zero. This is because the absolute value of the positive D.C. output of the rectifier 11 is substantially equal to that of the negative D.C. output of the rectifier 9 when there is no ambient noise. Thus, the loudspeaker 5 generates sound the volume of which is proportional to the ambient noise component in the output of the adder-amplifier 118. When the ambient noise ceases under this condition, the output of the inversion adder-amplifier 118 is reduced substantially to zero. If the analog switch 124 is conductive, the capacitor 139 is discharged through the resistor 137, and the input to the buffer 138 becomes a zero potential. Thus, no current flows through the light-emitting diode 142. The resistance of the CdS element 40 is restored,

whereby the output volume of the loudspeaker 5 is brought back to the initial value.

Thirs, it will be described how the apparatus of FIG. 7 operates when there is an ambient noise of a high level. When the microphone 6 picks up the ambient noise, the output volume of the loudspeaker 5 increases very much. Then, the D.C. output level of the rectifier 11 becomes higher than the voltage of the level comparator 114. The output of the level comparator 114 becomes a positive potential, whereby the non-stable multivibrator 134 starts working, thus generating pulses having such a waveform as illustrated in FIG. 3. As a result, the NAND circuit 135 produces pulses having the same waveform as shown in FIG. 3. The width of the pulses during which the pulse level remains positive is sufficiently small with respect to repetition period and the time constant determined by the resistor 137 and the capacitor 139.

The analog switch 124 remains conductive during the positive potential period of the pulse and non-conductive during the negative potential period of the pulse. In response to the pulses the analog switch 124 is rendered alternately conductive and non-conductive. The potential across the terminals of the capacitor 139 is substantially equal to the output level of the inversion adder-amplifier 118 immediately before the analog switch 124 is rendered repeatedly conductive and non-conductive. When the analog switch 124 is non-conductive, the potential of the capacitor 139 is maintained. When the switch 124 is conductive, the potential of the capacitor 139 varies according to the output of the inversion adder-amplifier 118. Since the conductive period of the analog switch 124 is far shorter than the non-conductive period, the potential of the capacitor 139 changes very little in comparison with the variation of the output of the inversion adder-amplifier 118. Thus, every time the above-mentioned positive pulse is generated, the output of the adder-amplifier 118 is detected repeatedly at specific intervals, each time for a very short time. While the analog switch 124 remains conductive, the potential of the capacitor 139 controls slowly the constant current circuit in accordance with the output of the inversion adder-amplifier 118. Thus, the output level of the volume adjusting circuit 3 changes slowly since the current flowing through the light-emitting diode 142, i.e. the load of the constant current circuit does not change sharply. The masking of the ambient noise is therefore suppressed. As a result, the output volume of the loudspeaker 5 can be adjusted in a natural manner.

Suppose the analog switch 124 is not provided and that the inversion adder-amplifier 118 is connected directly to the resistor 127. Then, the microphone 6 fails to detect the ambient noise if the sound from the loudspeaker 5 is large enough to mask the ambient noise. This is equivalent to a drop of the ambient noise level. The volume of the loudspeaker 5 is therefor reduced to such extent that the ambient noise is not completely masked. Then, the microphone 6 detects the ambient noise thereby to increase the output volume of the loudspeaker 5 again. The sound volume reduction and sound volume increase are repeated, thus annoying the listeners very much. Further, if the analog switch 124 were rendered non-conductive by a large volume of the loudspeaker 5 and remained non-conductive, the output of the inversion adder-amplifier 118 would not be supplied to the buffer 138 even if the ambient noise lowers thereby to drop the output voltage of the adder-amplifier 118. The volume of the loudspeaker 5 would not therefore be reduced. To prevent such an undesirable phenomenon, the non-stable multivibrator 134, NAND circuit 135 and inverter 136 are provided at the stages succeeding the level comparator 114.

Now it will be described how the apparatus shown in FIG. 7 operates when the loudspeaker 5 generates no sound, thus rendering the output level of the rectifier 11 lower than the voltage of the level comparator 113. In this case, the output of the level comparator 113 becomes a negative potential and make both analog switches 112 and 119 non-conductive. When these switches 112 and 119 are rendered non-conductive, the input and output of the buffers 115 and 120 become a zero potential because the non-inversion inputs of these buffers 115 and 120 are connected to the ground through the resistor 116 and the resistor 121, respectively. Thus, the output of the inversion adder-amplifier 118 becomes a zero potential, and no current flows through the light-emitting diode 142. As a result, the output level of the volume adjusting circuit 3 is maintained at the initial value. The analog switch 119 remains non-conductive even if an ambient noise exists under these circumstances. The output of the rectifier 9 is not therefore supplied to the buffer 120, and no current flows through the light-emitting diode 142. Accordingly, the volume adjusting circuit 3 is maintained at the initial value.

Suppose the analog switch 119 were not provided and that the output of the rectifier 9 were connected directly to the input of the buffer 120. Then, if the loudspeaker 5 remains silent for a certain period of time and if an ambient noise exists udring this period of time, the rectifier 9 produces a negative D.C. voltage corresponding to the ambient noise. This negative D.C. voltage is supplied to the inversion adder-amplifier 118 through the buffer 120 and converted into a positive D.C. voltage by the adder-amplifier 118. The positive D.C. voltage thus obtained is supplied to the buffer 138 through the analog switch 124, thereby reducing the attenuation ratio of the volume adjusting circuit 3. The moment the loudspeaker 5 starts generating sound under this condition, its output volume inevitably grows too large.

The analog switch 112 may be dispensed with. But, without it, the output of the rectifier 11 would contain a positive D.C. voltage which is generated by a noise or the like in the sound circuit system and which is lower than the voltage of the level comparator 113. If this happens, the positive D.C. voltage is supplied to the inversion adder-amplifier 118 via the buffer 115 and is converted into a negative D.C. voltage. The output of the adder-amplifier 118, which is thus a negative potential, is supplied through the analog switch 124 and the resistor 137 and is charged in the capacitor 139. As a result, the automatic volume adjusting operation is delayed a little by a period of time during which the potential across the terminals of the capacitor 139 changes from a negative value to zero potential when the loudspeaker 5 starts generating sound.

The analog switches 112 and 119 are provided to avoid a delay in automatic volume adjusting operation and an unnecessary increase in the output volume of the loudspeaker 5.

As mentioned above, in the embodiment of FIG. 7, two level comparators are used to detect the level of a D.C. signal obtained from an output sound signal of the volume adjusting circuit. A D.C. signal the level of which is higher than the lower one of the voltages of

the two level comparators is taken from a D.C. signal corresponding to a composite sound consisting of the sound from the loudspeaker and an ambient noise, thereby removing the sound component from the composite sound. The output obtained by this subtraction is utilized to control the output volume of the loudspeaker. When the level of the D.C. signal obtained from an output sound signal of the volume adjusting circuit is higher than the higher one of the voltages of the two level comparators, the output obtained by said subtraction is held, the output volume of the loudspeaker is repeatedly controlled for a predetermined time at specific intervals, and then the output obtained by the subtraction is released, thus adjusting the output volume of the loudspeaker little by little. Further, when the level of the D.C. signal obtained from an output sound signal of the volume adjusting circuit is lower than the lower one of the voltages of the two level comparators, the output obtained by the subtraction becomes a zero potential, whereby the output volume of the loudspeaker is not adjusted at all.

As described above, according to this invention there is provided an automatic volume adjusting apparatus wherein the output volume of a loudspeaker is adjusted automatically according to the level of an ambient noise, whereby the volume of sound never becomes unstable even if the ambient noise is masked by too large a sound from the loudspeaker, nor does the output volume of the loudspeaker become momentarily too large when the louspeaker starts generating sound.

What is claimed is:

1. An automatic volume adjusting apparatus comprising:

- a volume adjusting device connected between a sound signal source and a loudspeaker;
- means for generating a first signal corresponding to the level of a sound signal supplied from the sound signal source to the loudspeaker;
- means for generating a second signal corresponding to the level of a composite sound which includes sound generated by the loudspeaker and ambient noise;
- means for processing the first and second signals so as to generate a third signal corresponding substantially to only the level of the ambient noise;
- means for sampling the third signal at predetermined intervals;
- means for holding the output of the sampling means for a predetermined period of time; and
- means for controlling, according to the output of the holding means, the level of the sound signal passing through the volume adjusting device.

2. The automatic volume adjusting apparatus according to claim 1, wherein said holding means has a relatively small write-in time constant and a relatively large readout time constant.

3. The automatic volume adjusting apparatus according to claim 2, further comprising means for detecting the level of a sound signal supplied from said sound signal source to said loudspeaker and means for preventing the supply of the first and second signals to the third signal generating means when the level detecting means detects that the level of the sound signal is substantially zero.

4. The automatic volume adjusting apparatus according to claim 1, further comprising means for detecting the level of the first signal and means for causing said holding means to hold the output of said sampling means for the predetermined period of time when the level of the first signal is higher than a predetermined value and for supplying the output of the third signal generating means to said volume adjusting device when the level of the first signal is lower than the predetermined value.

5. The automatic volume adjusting apparatus according to claim 4, wherein said means for detecting the level of the first signal includes a first level comparator which produces an output when the level of the first signal is higher than a first predetermined value and a second level comparator which produces an output when the level of the first signal is lower than a second predetermined value smaller than the first predetermined value; and further comprising means for supplying the output of said third signal generating means to said volume adjusting device when the level of the first signal is higher than the first predetermined value and lower than the second predetermined value, means for causing said holding means to hold the output of the sampling means for said predetermined period of time, according to the output of the first level comparator, and means for stopping, according to the output of the second level comparator, the volume adjusting operation effected by the output of said third signal generating means.

* * * * *

UNITED STATES PATENT AND TRADEMARK OFFICE
CERTIFICATE OF CORRECTION

PATENT NO. : 4,254,303

DATED : March 3, 1981

INVENTOR(S) : Tetsuya TAKIZAWA

It is certified that error appears in the above—identified patent and that said Letters Patent is hereby corrected as shown below:

<u>IN THE DRAWINGS</u>:

Fig. 7, change the algebraic signs of the comparators 113 and 114 from "$\frac{-}{+}$" to --$\frac{+}{-}$--.

Signed and Sealed this

Second **Day of** *June 1981*

[SEAL]

Attest:

RENE D. TEGTMEYER

Attesting Officer *Acting Commissioner of Patents and Trademarks*

HPS Trailer Page
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UNITED STATES PATENT AND TRADEMARK OFFICE

CERTIFICATE OF CORRECTION

PATENT NO. : 5,608,410

DATED : March 4,1997

INVENTOR(S) : Louis A. Stilp, Curtis A. Knight and John C.Webber

It is certified that error appears in the above-indentified patent and that said Letters Patent is hereby corrected as shown below: On the title page, and in

Col. 1, in the title delete "CROSS REFERENCE TO RELATED APPLICATIONS".

Col. 13, line 65, the word "Calculated" delete the capital "C".

Col. 14, line 64, the word "AS" delete the capital "S".

Col. 21, line 43, change "ω" to --Ω--

Col. 26, line 16, "stems" should be --steps--.

Signed and Sealed this

Twenty-second Day of July, 1997

Attest:

BRUCE LEHMAN

Attesting Officer

Commissioner of Patents and Trademarks

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Document Listing

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US003849603	15	1 - 15	1
Total (1)	15	-	-

United States Patent [19]

Proios

[11] **3,849,603**

[45] **Nov. 19, 1974**

[54] **REMOTE BANKING INTERCOM SYSTEM**

[75] Inventor: **Steve Proios**, Northvale, N.J.

[73] Assignee: **The Mosler Safe Company**, Hamilton, Ohio

[22] Filed: **Oct. 30, 1972**

[21] Appl. No.: **302,253**

[52] **U.S. Cl.** .... **179/1 H,** 179/1 VC

[51] **Int. Cl.** **H04m 1/00**

[58] **Field of Search** .... 179/1 VC, 1 H, 1 B, 1 HF, 179/81 B, 1 CN, 1 P

[56] **References Cited**

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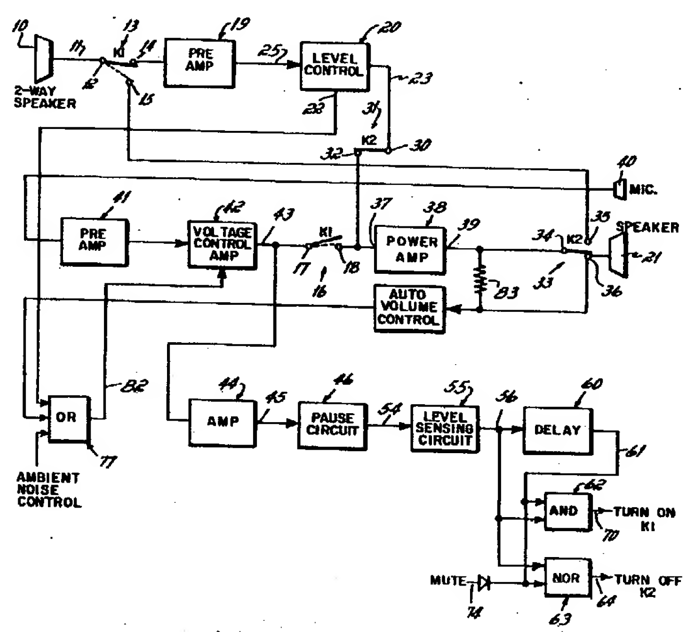
Primary Examiner—Ralph D. Blakeslee
Assistant Examiner—Bradford Leaheey

[57] **ABSTRACT**

An intercom system for remote banking which facilitates communication between a teller terminal and a remotely located customer terminal. A two-way speaker is provided at the customer's location and a separate microphone and a speaker are provided at the teller's location. An amplifier in the communication path linking the two stations is connected such that normally its input is responsive to the customer two-way speaker with its output feeding the teller speaker, permitting the customer to talk to the teller. When the teller speaks into the microphone, and assuming his volume level exceeds a predetermined first level, the amplifier automatically switches to connect the teller microphone to the amplifier input and the customer two-way speaker to the amplifier output, permitting the teller to talk to the customer. When the level of the teller microphone signal falls below a second predetermined level which is less than the first level to permit the teller's voice to drop without losing control of the direction of communication, the intercom system will switch to again permit the customer to talk to the teller. The system is additionally provided with a muting signal which disconnects the teller speaker whenever the ambient noise level at the customer's location is high due to mechanical movement of a drawer or pneumatic carrier at the customer's location.

16 Claims, 6 Drawing Figures



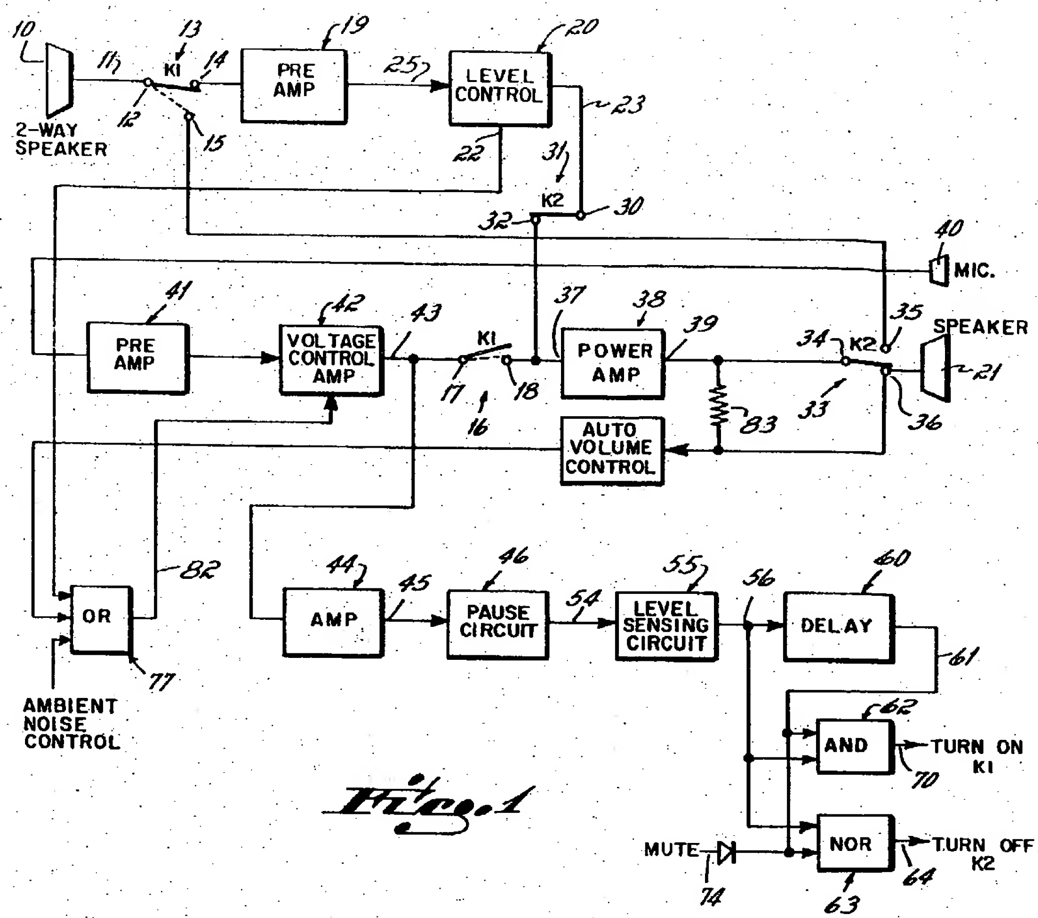

2-WAY SPEAKER
K1
PRE AMP
LEVEL CONTROL
K2
MIC.
PRE AMP
VOLTAGE CONTROL AMP
K1
POWER AMP
K2
SPEAKER
AUTO VOLUME CONTROL
OR
AMBIENT NOISE CONTROL
AMP
PAUSE CIRCUIT
LEVEL SENSING CIRCUIT
DELAY
AND
TURN ON K1
MUTE
NOR
TURN OFF K2

Fig. 1

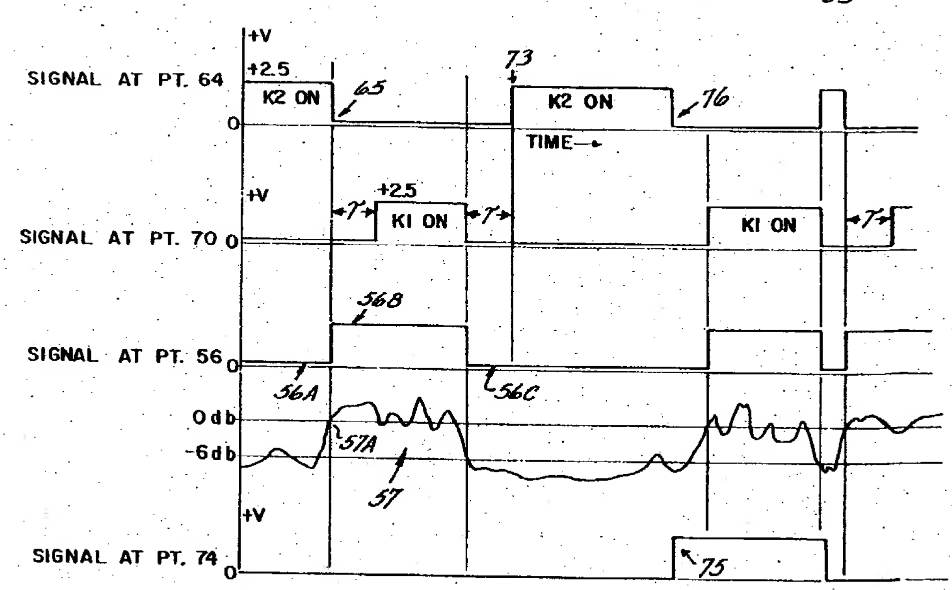

SIGNAL AT PT. 64
+V
+2.5
K2 ON
K2 ON
TIME
SIGNAL AT PT. 70
K1 ON
K1 ON
SIGNAL AT PT. 56
0 db
-6 db
SIGNAL AT PT. 74

Fig. 2

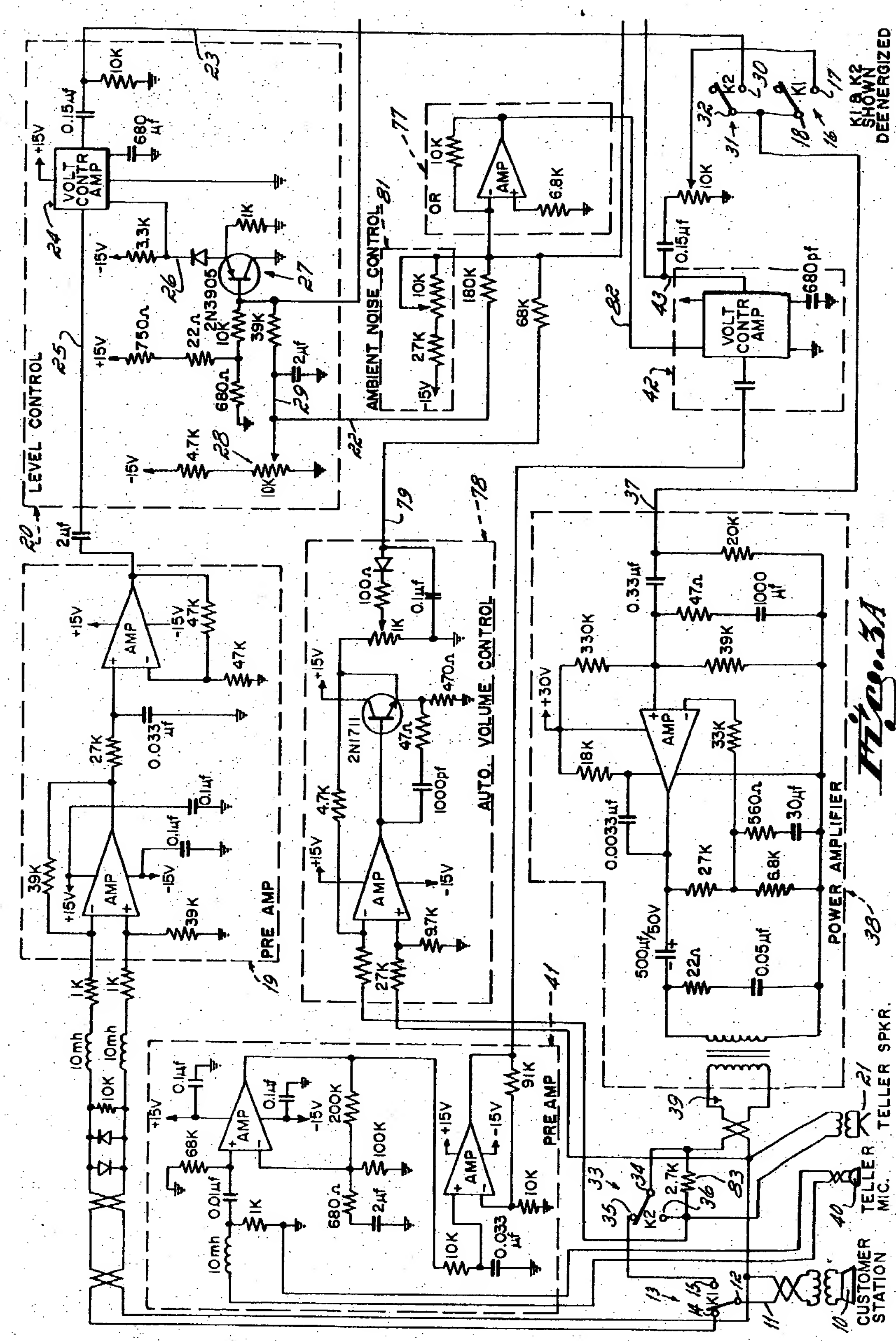

LEVEL CONTROL
VOLT CONTR AMP
2N3905
AMBIENT NOISE CONTROL
OR
AMP
PRE AMP
AUTO. VOLUME CONTROL
2N1711
POWER AMPLIFIER
K1 & K2 SHOWN DEENERGIZED
TELLER SPKR.
TELLER MIC.
CUSTOMER STATION

FIG. 3A

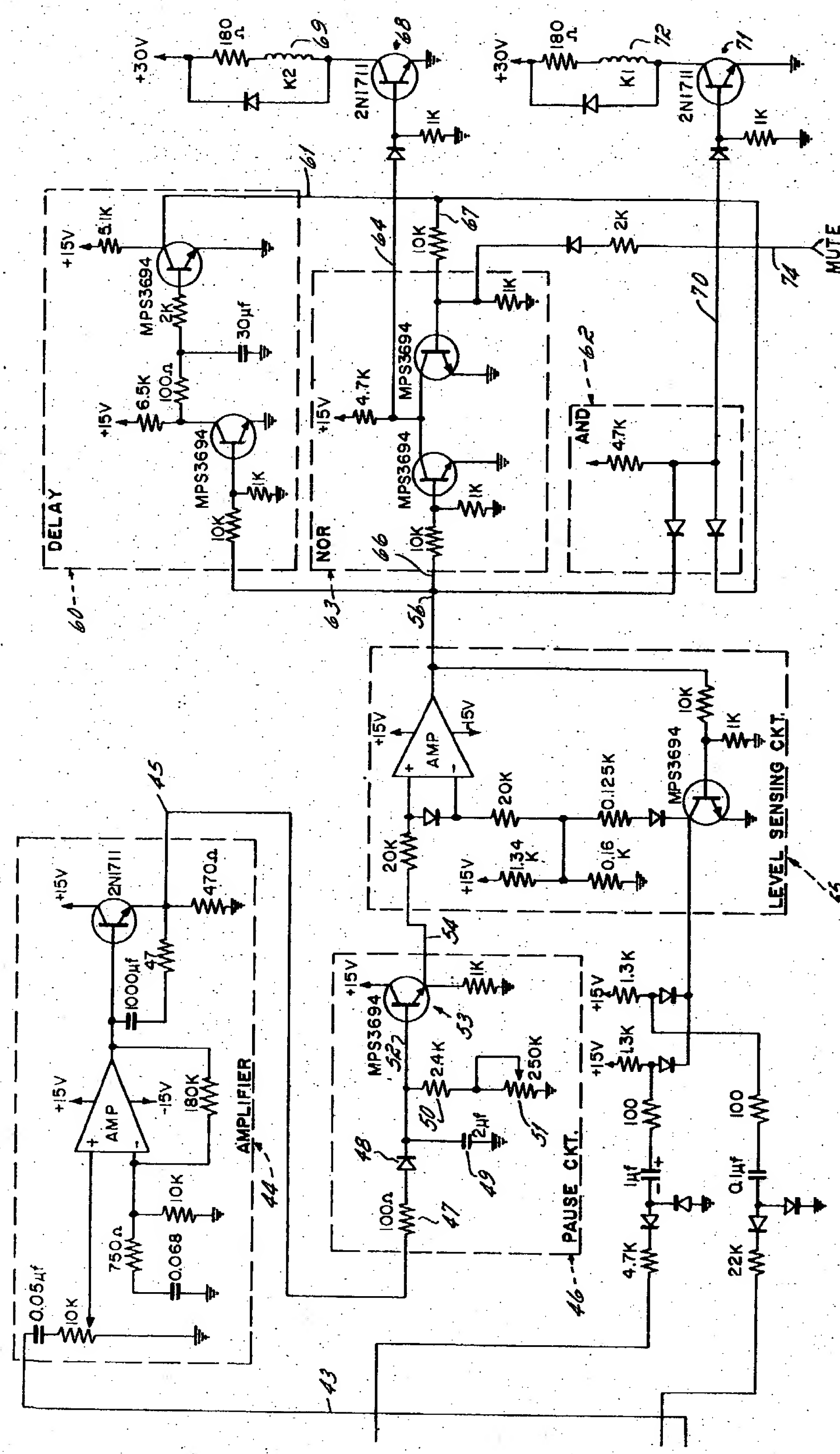

DELAY
NOR
AND
MUTE
LEVEL SENSING CKT.
PAUSE CKT.
AMPLIFIER
AMP
MPS3694
2N1711
K1
K2
+30V
+15V
-15V

Fig. 3B

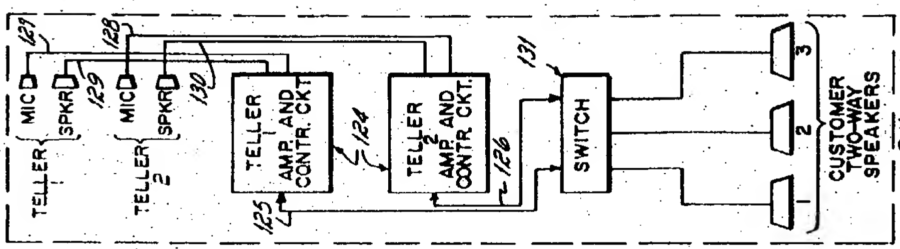

TELLER 1
MIC
SPKR
TELLER 2
MIC
SPKR
TELLER 1 AMP. AND CONTR. CKT.
TELLER 2 AMP. AND CONTR. CKT.
SWITCH
CUSTOMER TWO-WAY SPEAKERS
127
128
129
130
131
124
125
126
1
2
3

Fig. 4

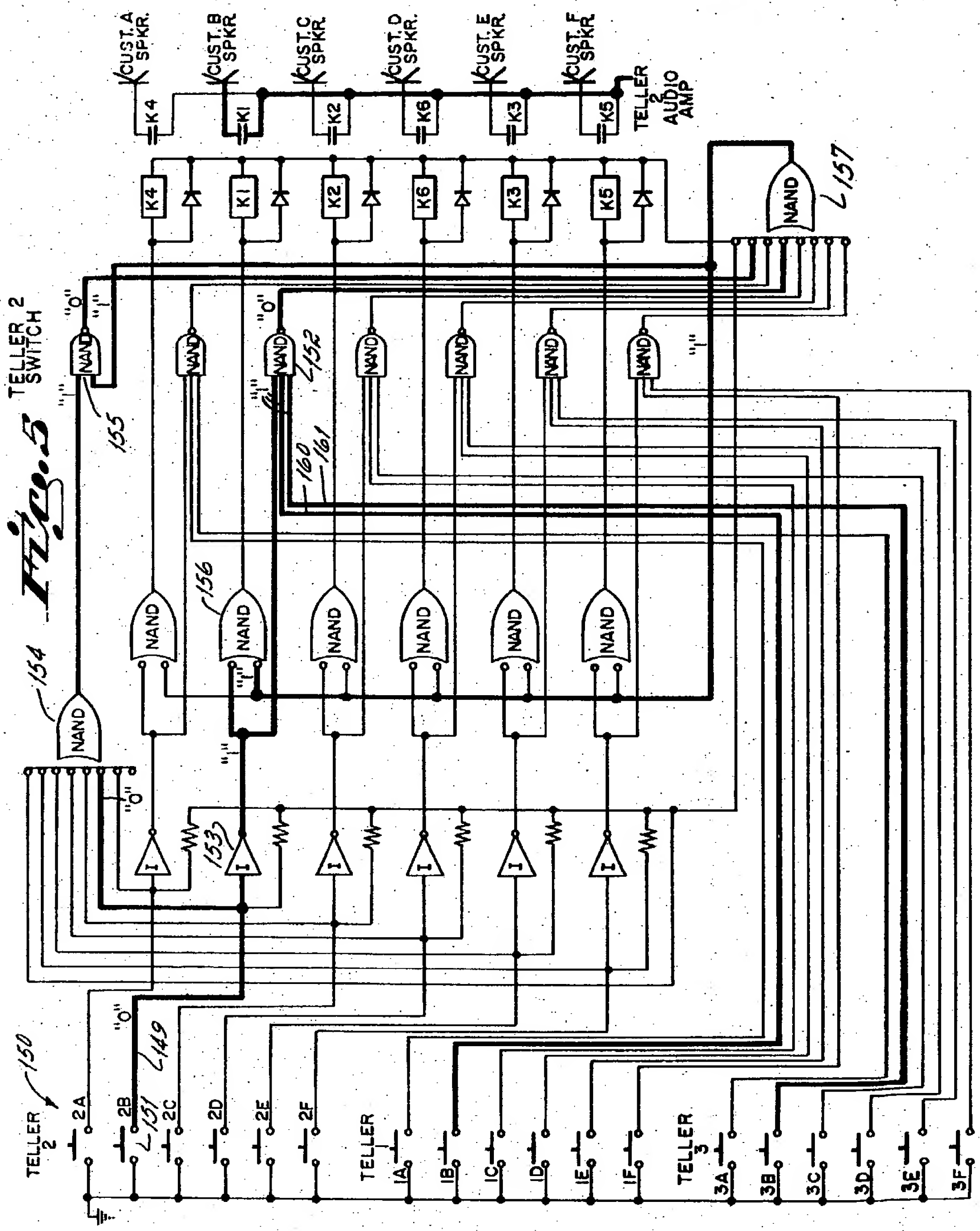

CUST. A SPKR.
CUST. B SPKR.
CUST. C SPKR.
CUST. D SPKR.
CUST. E SPKR.
CUST. F SPKR.
TELLER 2 AUDIO AMP
K4
K1
K2
K6
K3
K5
NAND
TELLER 2 SWITCH
"0"
"1"
150
151
149
152
153
154
155
156
157
160
161
TELLER 2
2A
2B
2C
2D
2E
2F
TELLER 1
1A
1B
1C
1D
1E
1F
TELLER 3
3A
3B
3C
3D
3E
3F

Fig. 5

REMOTE BANKING INTERCOM SYSTEM

This invention relates to intercom systems and more particularly to a remote banking system intercom for communicating between a bank teller location and a remote customer location.

In the drive-in banking field, a form of remote banking, the bank teller and customer are usually separated by a distance anywhere from 6 to 30 feet, with the customer being in his car and the teller in a fully-enclosed booth attached to the main bank building which has a window looking out on the customer terminal. There may be a single teller station uniquely associated with one customer unit; or alternatively several teller stations servicing an even larger number of customer units, with each teller being able to service any one of the customers. Such drive-in systems, whether multistation or not, permit the customer to conduct banking business from his car. In a related type of remote banking system, a sidewalk window is provided where customers, albeit not in cars, conduct their business from the sidewalk without having to enter the bank.

Remote banking systems of the types described require some method for transferring cash, checks, deposit slips and the like between the teller and the customer. In one form, a sliding drawer, controlled by the teller, is provided. The drawer is opened proximate the customer to permit the customer to deposit a check or cash therein. The teller then closes the drawer and thereafter removes the articles from the drawer. The teller then performs certain operations for accounting purposes before placing a receipt, cash or the like in the drawer. The drawer is again opened to the customer to allow the customer to remove the item from the drawer. While a sliding drawer is commonly used, a pneumatic carrier conveying system is also used, the pneumatic carrier system taking the place of the sliding drawer. An advantage of the pneumatic carrier approach is that it permits the teller to be located large distances from the customer. Pneumatic systems are particularly adaptable to drive-in or walk-up banking systems having many customer locations serviced by one or more tellers.

In all remote banking systems an intercommunication system is necessary to permit conversation between the teller who is in an enclosed booth, and the customer who may be many feet away. The customer must be able to inform the teller of any special request he may have, for example, the type of change he would prefer, and the teller has to be able to speak to the customer, for example, to ask the customer to endorse a check.

Experience has shown that it is preferable to place the teller in control of the direction of communication for the intercom since it is the teller who most often must initiate conversation. Additionally, for ease of operation the communication system should operate with as little manual intervention as possible to permit the teller and the customer to use their hands for other purposes. Accordingly, by permitting the teller's voice to control the direction of communication, not only will the teller be in control, but the teller's hands will be free to write or to use them for some other transaction-related purpose.

An intercommunication system for a remote banking installation must also function in a manner which accommodates varying conditions of use. For example, while the teller is speaking to the customers, he may have to step back away from the microphone in order to use an adding machine, view cash he is removing from the cash drawer, or the like. While this causes the volume of the teller's voice to drop, it should not cause the communication direction, which is under control of the teller's voice, to switch until the teller has finished talking.

Another variable in a remote banking system is the volume of the customer's voice as reproduced by a speaker at the teller's location. This sound will couple into the teller's microphone, but should not activate the intercom direction switching controls. While the system must reproduce the normal customer's voice at a level loud enough for the teller to hear, means must be provided to prevent the customer's voice from activating the communication direction switching controls should the customer suddenly speak loudly.

A further variable in a remote banking system arises from noise caused by the mechanical movement of the drawers an/or pneumatic carriers or caused by other ambient noise at the customer's location which occurs when the customer is talking. If these customer station noises, when reproduced at the teller station, are picked-up by the teller's microphone, the direction of communication will be reversed, cutting-off the customer.

It is a primary object of this invention to provide an intercommunication system for remote banking installations which permits only the teller to control the direction of communication with his varying amplitude voice while preventing annoyance caused by switching and mechanical sounds.

It is a related object of this invention to prevent customer station-originated noise reproduced at the teller's speaker from switching the direction of communication while the customer is still speaking.

It is still another related object of this invention to prevent the mechanical sounds of drawer opening and pneumatic carrier operation from being reproduced at the teller's location, coupling back into the teller's microphone and reversing the communication direction.

It is another object of this invention to provide an intercom system having a switchable communication direction controlled by the volume of the teller's voice. When the teller's voice rises above a first threshold, the teller is permitted to talk to the customer and limited variation in level of the teller's voice below this threshold is permitted thereafter without changing the communication direction.

These and other objects and advantages are achieved by a remote banking intercom system which includes a two-way speaker at the customer's location which is normally connected to the input of a power amplifier whose output is connected to a speaker at the teller's location,, permitting the customer under normal conditions to talk to the teller. A microphone at the teller's location is provided which is connected to an amplitude-sensing circuit for controlling the direction of communication. When the sensed level at the teller microphone exceeds a first level, the two-way speaker at the customer location is switched from the amplifier input to the amplifier output and the microphone at the teller location is connected to the amplifier input, permitting the teller to talk to the customer. Communication from the teller microphone to the customer two-

way speaker continues as long as the volume at the teller microphone does not fall below a second predetermined level, which is less than the first level. Thus, the volume of the teller's voice can vary within limits without effecting reversing the direction of communication. Of course, if the level at the teller microphone drops below the second threshold, as occurs when the teller stops talking, the system will revert to its normal condition with the customer being able to talk to the teller.

The intercom system of this invention further includes a microphone sensitivity control network for reducing the sensitivity of the microphone under certain conditions. The sensitivity reduction occurs whenever the system is connected to permit communication from the customer to the teller, and operates to prevent a customer, speaking in a loud tone of voice, from switching the communication direction via audible feedback from the teller speaker to the teller microphone. Preferably, the microphone sensitivity is also reduced under conditions of high ambient noise at the teller's station to prevent ambient noise conditions at the teller's location from switching the direction of communication when a customer is talking. In addition, a muting circuit is provided for disconnecting the teller's speaker whenever an interconnected mechanical apparatus, such as a sliding drawer or pneumatic carrier, will produce a high noise level at the customer's location which noise when reproduced at the teller speaker and coupled back to the teller microphone would switch the communication direction.

The foregoing and other objects, features and advantages of this invention will be more readily understood from the following more detailed description of a preferred embodiment thereof taken in conjunction with the accompanying drawings which form a part of this disclosure wherein:

FIG. 1 shows a schematic diagram, in block circuit format, of a preferred embodiment of this invention;

FIG. 2 is a timing diagram showing signals at various points in the schematic diagram of FIG. 1;

FIGS. 3A and 3B are detailed circuit diagrams of one form of the system shown in FIG. 1;

FIG. 4 shows schematically a remote banking intercom system with two teller and three customer locations; and

FIG. 5 shows a switching network for a remote banking intercom system with more than one teller location and also more than one customer location.

Referring now to FIG. 1, a schematic diagram is shown of an intercommunication system particularly adapted for use in a remote banking system. A two-way speaker **10** is provided at the customer location and operates as a microphone for communications from the customer to the teller, while operating as a speaker for communications from the teller to the customer.

An electrical wire **11** is provided to electrically connect the customer two-way speaker **10** to the input terminal **12** of a single-pole, single-throw switch **13**. The electrical wire **11** may be any type of electrical cable having a low d.c. resistance, although a shielded twisted pair or other form of shielded cable is advantageous in reducing stray coupling of spurious electrical signals which may be present in systems where the customer is located at a distance from the teller.

The single-pole, double-throw switch **13** has in addition to the input terminal **12** two output terminals **14** and **15**. The switch **13** may take the form of an electronic switch, or, in a preferred embodiment, may be a relay (K1) activated contact, for electrically connecting the input **12** to one of the two outputs **14** and **15**.

A single-pole, single-throw switch **16** with an input terminal **17** and an output terminal **18** is also provided in the network and is switched at the same time as switch **13**. The switch **16** may also comprise an electronic switch having the same control signal as switch **13**, or may comprise a separate set of contacts controlled by the relay **K1**. Because switches **13** and **16** operate together, the input **12** of switch **13** is connected to the output **14** at the same time as the input **17** of the switch **16** is disconnected from the output **18**. When the switch **13** is changed, the switch **16** will also be changed so that the input **12** for switch **13** connects to the output **15** while the input **17** of switch **16** connects to the output **18**.

The switch **13** has the primary function of switching the electrical connection to the customer two-way speaker **10** to permit the customer to either talk or listen. In the customer talk mode, switch **13** connects the input **12** to the output **14** permitting electrical signals from the two-way speaker **10** to be amplified by a preamplifier circuit **19**. In the customer listen mode, switch **13** connects the input terminal **12** to the output terminal **15** permitting the two-way speaker **10** to be connected to a power amplifier **38** and allow the customer to hear the teller's voice.

In the preferred embodiment shown in FIGS. 3A-3B, the preamplifier **19** comprises two series-connected integrated circuit amplifiers. While an integrated circuit amplifier has been utilized, a discrete component amplifier circuit with similar gain characteristics may also be used without degrading system performance.

The output of the preamplifier **19** is connected directly to the input of a level control means **20** which provides the teller with a volume control for the teller speaker **21**. The level control **20** may take the form of a variable resistor or some other conventional means for varying the level of a signal. The level control means **20** has an output wire **22** for carrying a control signal whose amplitude depends upon the setting of the level control means **20**. This level-indicating signal is utilized by feedback control logic to be described later.

The level control means **20** includes circuitry for controlling the output level of the audio signal on the output wire **23**. In a preferred embodiment, the level control means **20** advantageously comprises a voltage controlled amplifier that has a controllable variable gain characteristic responsive to a control voltage. In FIG. 3A the voltage-controlled amplififer **24** has an input lead **25** connecting to the control terminal of the voltage-controlled amplifier. The output terminal of voltage-controlled amplifier **24** is connected through a high pass filter to the output line **23**. The control signal which is applied to the control terminal of the voltage control amplifier **24** is generated by varying the voltage at point **26**. The varying voltage at point **26** is produced by controlling the current through transistor **27**. This varying current is produced by the adjustable biasing network connected to the base of transistor **27** which includes a variable resistor **28**, normally remotely located at the teller's location to permit volume adjustment. By varying the voltage on the wire **29**, the cur-

rent through transistor 27 can be varied, producing a varying voltage at the control input to the voltage control amplifier 24. The level-carrying wire 22 connects directly to the wire 29 and carries a voltage which is related to the output amplitude of the voltage-controlled amplifier 24.

Referring again to FIG. 1, the output signal on line 23 connects to the input terminal 30 of a single-pole, single-throw switch means 31, the switch 31 being normally closed (K2 on) allowing the signal at the input 30 to be connected directly to the output terminal 32. Another switch means 33 of the single-pole, double-throw type having an input terminal 34 and two output terminals 35 and 36 is provided and is designed to switch at the same time as switch means 31. Normally switch means 33 will connect the input terminal 34 to the output terminal 36 and at the same time, switch means 31 will connect the input 30 to the input 32. When switch means 31 is changed, switch means 33 will also be changed so that the input terminal 30 is disconnected from the output terminal 32 of switch means 31 and the input terminal 34 of switch means 33 is connected to the output terminal 35.

Switch means 31 and 33 may take the form of electronic switches with common control signals, or advantageously may take the form of contacts of a relay K2.

The amplified signal of the customer's voice is carried by a wire from the output terminal 32 of switch means 31 to the input terminal 37 of the power amplifier 38. The power amplifier 38 shown in the preferred embodiment in FIG. 3A is an integrated circuit amplifier, although a discrete component amplifier circuit is equally acceptable. The amplifier output 39 connects directly to the input terminal 34 of switch means 33 and, when switch means 33 is in its normal position (FIG. 1), the amplifier output signal will be connected to the teller's speaker 21. When the relays K1 and K2 are in the normal position shown in FIG. 1 (K1 off and K2 on), the two-way speaker 10 is connected in a talk mode via switch means 13 to the preamplifier 19, the level control means 20, the power amplifier 38 and the speaker 21 permitting the customer to talk to a teller at a remote location.

The teller microphone 40 and the teller speaker 21 are normally located close to each other at a location which is remote from the customer location. The teller microphone 40 is connected to a preamplifier circuit 41 which, in the preferred embodiment shown in FIG. 3A, comprises two series-connected integrated circuit amplifiers. While integrated circuit amplifiers are preferred, other discrete component amplifiers are equally acceptable as long as the amplifier gain is sufficient to provide adequate volume at the customer location to allow the teller to be heard by the customer.

The output of the preamplifier 41 is connected by a wire to the input of a voltage control amplifier circuit 42. The voltage-controlled amplifier 42 controls the microphone gain to prevent feedback from the teller speaker 21 to the teller microphone 40, a feature which will be explained later in greater detail. The output of the voltage-controlled amplifier 42 appears on line 43 and is connected by a wire to the input terminal 17 of switch 16 and to the input terminal of the amplifier 44 which includes, in a preferred embodiment, an integrated circuit amplifier followed by a discrete component driver circuit to produce an amplified teller's signal on the output line 45.

The amplified output of the teller's voice on wire 45 is applied to the input of a pause circuit 46. As shown in FIG. 3B, pause circuit 46 includes a resistor 47 connected at one end to the output line 45 and at its other end to a diode 48, the diode 48 providing a half-wave rectification of the amplified teller's voice. The output of the diode 48 is connected to a capacitor 49 which becomes charged whenever a signal from the teller's microphone is applied to the input of the pause circuit. Connected in parallel with the capacitor 49 is a series-connected resistor 50 and a variable resistor 51 which provide a variable resistance discharge path for the capacitor 49. The diode, the positive lead for the capacitor 49 and the series-connected resistors all connect to the base lead 52 of a transistor 53 which forms an emitter-follower circuit producing an output for the pause circuit at line 54.

In operation the pause rectifies the teller speech signal received on line 45 from amplifier 44. The positive voltage peaks build up a positive charge on the capacitor 49. When the output of the amplifier 44 has a voltage which is less than the voltage across the capacitor 49, the capacitor will discharge gradually through resistors 50 and 51. Since the charge remains on the capacitor 49 after the teller stops talking, the retained charge is used by a level sensing circuit (described later) to maintain control over the direction of communication and permit the teller to pause without changing the communication direction.

The output of the pause circuit 46 is on wire 54 which is connected to the input of a level-sensing circuit 55. The level-sensing circuit 55 is an adjustable though preset circuit which is designed to produce a signal at one level only when the input exceeds a first predetermined reference level and to produce an output at a second level only when the input falls below a second predetermined level which is less than the first predetermined reference level. The output of the level-sensing circuit 55 appears at output point 56 and a typical output signal is shown in FIG. 2 for the input at line 54. With reference to FIG. 2, when the input level of the teller speech signal on wire 54 exceeds an arbitrarily selected reference level shown as zero db. at 57A, the output of the level-sensing circuit 55 shifts from a first level shown at 56A to a second level shown at 56B. Once the output of the sensing circuit 55 switches to the second level, the speech input signal 57 on wire 54 can vary above or below the 0 db. reference level. If the speech signal level at input 54 falls to −6db., the sensing circuit 55 output 56 returns to its first level 56C. By permitting the output of level-sensing circuit 55 to remain constant over a wide range of input amplitudes as will be seen shortly, the teller can maintain control over the communication direction even if the amplitude of his voice at the teller microphone varies above and as much as 6 db. below the arbitrarily selected zero db. reference level required to activate teller communication to the customer.

The level-sensing circuit 55 as shown in FIG. 3B for a preferred embodiment of the present system comprises a Schmitt trigger circuit. It is well known that a Schmitt trigger will produce an output at one level whenever the input signal exceeds a predetermined level which is adjustable by controlling the trigger bias. It is a further known characteristic of Schmitt triggers

that the output is unchanged until the input level falls a predetermined amount below the input level required to produce the switched output. Consequently, a Schmitt trigger circuit is ideally suited for the application in the intercom system because it will permit the teller input signal to drop a predetermined amount without changing the output. While a Schmitt trigger circuit has been shown in FIG. 3B as a preferred embodiment for the level-sensing circuit **55**, it will be recognized by those of skill in the art that certain other circuit configurations may also be utilized to produce the same results achieved with a Schmitt trigger.

Because the output signal from the level-sensing circuit **55** is a two-level signal having an up and a down level, this output signal is ideally suited for driving digital logic circuitry to control the switches **13, 16, 31** and **33.** The signal at point **56** is applied to the input of a delay circuit **60,** the delay output **61** being a signal identical to the input at point **56** delayed by a delay time τ. The delay circuit **60** may take the form of a digital delay line or, as in a preferred embodiment shown in FIG. 3B, may take the form of two amplifier stages with a capacitor in the base circuit of the second amplifier stage whose charging and discharging charging through a resistor is utilized to delay the output on line **61** by a time related to the RC time constant. The delayed signal on line **61** is applied to one input of an AND circuit **62** and one input of the NOR circuit **63.** The second input to AND circuit **62** is the signal on line **56.** This signal is also applied to a second input to the NOR circuit **63.**

In operation, the NOR circuit **63** has an output **64** which is approximately 2.5 volts for the preferred embodiment shown in FIG. 3B when the input on line **56** is approximately zero volts, a condition which indicates that the volume at the teller microphone is below the arbitrary reference level designated 0 db. However, when the input signal on line **56** rises to a positive value as depicted at point **56B** in FIG. 2, the output **64** of the NOR circuit **63** falls to substantially 0 volts as shown generally at **65.** In fact, if the voltage at either NOR input is positive, the output **64** of the NOR circuit **63** will be substantially zero volts. If both inputs **66** and **67** have a zero or negative voltage applied thereto, the output **64** will be approximately +2.5 volts.

The output **64** of NOR circuit **63** is coupled directly to the base of the K2 relay driving transistor **68.** Whenever the output **64** of the NOR circuit **63** is positive, transistor **68** will be conducting and current will pass through the K2 relay coil **69** causing the relay contacts to be closed. Switches 31 and 33 are controlled by the K2 relay and are drawn in FIG. 1 with the electrical connections corresponding to the conditions when the K2 relay coil **69** has a current flowing therethrough, i.e., K2 is on. When the voltage at the NOR output **64** falls to zero, however, transistor **68** will no longer conduct and the K2 relay will be de-energized, thus changing the electrical connections for switches **31** and **33.**

The AND circuit **62** will produce a positive output signal on line **70** whenever both inputs are at a positive level, and will have a zero voltage if either input is zero. Since one of the inputs is the signal on line **56** and the other input is the signal on line **56** delayed by a time delay τ, the output of AND circuit **62** will rise to a level of approximately +2.5 volts only when both inputs have become positive. Consequently, as seen in FIG. 2, the output **70** rises to its positive level a time period τ after the signal on line **56** rises which corresponds to the same time delay after the signal on line **64** falls to its substantially zero voltage value.

The signal on line **70** is connected directly to the base circuit of the K1 relay driver transistor **71.** Whenever the signal on line **70** is positive, the transistor **71** will conduct and cause a current to flow through the K1 relay coil **72.** The switching of the K1 relay will cause switches **13** and **16** of FIG. **1** to be changed from the configuration there shown to the connection indicated in dotted lines.

Because of the sequential nature of the logical elements used for controlling the K1 and K2 relays, these relays operate in a predetermined sequence to switch the system configuration. In the system in FIG. **1,** whenever the teller is silent and no signals are being received by the teller microphone **40,** the K1 and K2 relays are positioned as shown and permit a customer to speak into the two-way speaker **10** and talk to the teller. For the configuration which permits the customer to talk to the teller, the K2 relay is energized and the K1 relay is not energized. When the teller speaks, however, the system configuration will be changed sequentially. This is best shown in FIG. **2** at the point **57A** where the teller's voice is of a sufficient volume to begin system reconfiguration. Whenever the teller's voice exceeds the predetermined level indicated as 0 db., the signal at line **64** will fall to its down level as indicated at point **65** to cause the K2 relay to turn off and change the switches **31** and **33.**

When the K2 relay is de-energized, the output of the level control means **20** is disconnected from the switch output terminal **32,** consequently removing the two-way speaker **10** from the input to the power amplifier **38** and preventing customer signals from reaching the input to the power amplifier **38.** Simultaneously, switch **33** is changed so that the output **39** from the power amplifier **38** is connected electrically to the output terminal **35** of the switch **33,** disconnecting the speaker **21** from the power amplifier **38.**

After the time period τ has elapsed, the K1 relay is activated and the switch **13** is changed to connect the speaker **10** to the output of the power amplifier **38.** Simultaneously, the switch **16** is also activated which connects the microphone **40** to the input to the power amplifier **38** which completes the necessary system reconfiguration to permit the teller to speak into the microphone **40** and have the customer hear the teller's voice reproduced by the two-way speaker **10.**

When the teller stops talking for a time period greater than is required for the discharge of the capacitor **49** in the pause circuit **46,** or if the teller speaks at a volume level substantially reduced, i.e., 6 db.'s below the triggering level, the signal at point **56** will again change and initiate a sequential switching of the relays K1 and K2. As shown in FIG. **2,** when the teller signal at point **56** falls to the level indicated generally at **56C,** the signal at point **70** immediately falls to its substantially zero voltage level, causing the K1 relay to de-energize and change the switches **13** and **16** back to their original position as shown in FIG. **1.** A time period τ later, the NOR input **67** will have fallen to its down level which causes the K2 relay to be energized by the rise of the signal on line **64** to its value of approximately +2.5 volts as indicated at point **73.**

The switching reconfiguration when the teller's voice ceases, first operates to disconnect the microphone **40** from the input of the power amplifier **38** and connect the customer two-way speaker **10** to the preamplifier **19**. This early switching of the two-way speaker **10** to the preamplifier **19** is highly advantageous. Delayed by a time period τ after the K1 relay has been switched, the K2 relay is energized causing switches **31** and **33** to change to the configuration shown in FIG. **1** again permitting the customer to speak into the two-way speaker **10** and the teller to hear the customer through the teller speaker **21**.

The sequential switching just described is highly advantageous to the operation of the system. When the configuration is being changed from the customer talk mode to the teller talk mode, the teller speaker **21** is disconnected from the power amplifier **38** at the same time as the customer signals are removed from the power amplifier input **37**. This removing of the teller speaker **21** from the amplifier output **39** prevents the subsequent switching of switch **13** from causing a clicking sound to be heard by the teller. Similarly, the removal of the customer two-way speaker **10** from the power amplifier output prior to the connecting of the teller speaker **21** to the power amplifier output and also prior to completing the connections required in the customer talk mode will prevent a blast of sound from being heard by the teller. The customer two-way speaker **10** will have stored energy therein when the teller stops talking. If the system were immediately reconfigured, this stored energy would be amplified and reproduced by the teller speaker **21**. By connecting the customer two-way speaker **10** to the preamplifier before the teller speaker is connected to the amplifier, the stored energy is dissipated which operates to eliminate any blast of sound in the teller speaker **21** when the system reconfigured from the teller talk to the customer talk mode.

The intercom system of this invention is designed primarily for application in remote banking systems which characteristically have mechanical drawers or pneumatic carriers being moved to permit papers to be transmitted between the customer and the teller. At the customer's location, the pneumatic carrier or drawer is typically located near the two-way speaker. Because these moving elements of the remote banking system generally produce a great deal of noise, it becomes important to prevent this noise from being transmitted to the teller by the intercom system. To prevent the noise associated with mechanical movement at the customer's location from being transmitted to the teller, a muting circuit is provided which includes an input wire **74** which is normally maintained at a substantially zero voltage level. When the teller activates a moving drawer or a pneumatic carrier at the customer's location, a positive muting signal is applied to the muting input **74** which is shown at **75** in FIG. **2**. The positive muting signal can be generated in many ways. For example, when a teller-actuated drawer is moved by the teller, a switch can be activated by the drawer mover mechanism to apply a positive signal on the muting signal input line **74** during the drawer movement. In pneumatic systems, the signal can be generated whenever the pneumatic switch associated with the customer location is set to permit a pneumatic carrier to arrive at or leave from a customer location. This positive muting signal at input **74** is transmitted to the input **67** to NOR circuit **63** and causes the output **64** to drop as shown at **76**. The muting signal, therefore, causes the K2 relay to turn off, a condition which disconnects the speaker **21** from the power amplifier **38** to prevent sounds at the customer's location from being amplified and heard at the teller's location.

The muting signal is also applied to the AND circuit **62** which permits the switching of the K1 relay whenever the output **56** of the level-sensing circuit **55** becomes positive. This permits the teller to speak to the customer if he so desires while the muting signal is present. Communications from the customer to the teller, however, are prevented until the muting signal is removed, a condition which will occur when the mechanical movement of the drawer or pneumatic carrier has been completed.

In intercommunication systems of the type adapted for use in remote banking installations, it is important that the direction of communication be controlled by the teller. Since signals applied to the teller's microphone **40** are used to generate the logic control signal at point **56**, it is important to prevent sounds produced by the speaker **21** from being coupled to the microphone **40** and cause the network to switch when such switching is not desired. For example, a customer might have a very loud voice and it is important to prevent the customer's voice, as reproduced by the speaker **21**, from switching the control relays and cut off the customer's communication with the teller. Further, since the teller can control the level of the customer's voice by the level control circuit, it is desirable to prevent an increased customer volume from switching the direction of communication. Another source of undesirable switching is from the ambient noise conditions at the teller location itself. These noises arise from various activities within the bank itself and, as such, should not activate the switching logic to permit the teller to communicate with the customer.

To prevent the various feedback noises from switching the direction of communication of the intercom system, a feedback control system is included in the intercom system shown in FIG. **1**. The signal-carrying wire **22** has a voltage thereon which is proportional to the teller's setting of the variable resistor **28** which controls the volume of the customer's voice as reproduced by the speaker **21**. This signal is applied directly to the analog summing circuit **77**.

To compensate for the feedback generated by the sound reproduced by the speaker **21**, an automatic volume control amplifier **78** is electrically connected to the speaker **21** for receiving amplified electrical signals coming from the customer's location. The automatic volume control amplifier **78** preferably comprises an integrated circuit amplifier with an output wired to a variable gain amplifier stage whose output **79** is electrically connected to one input of the analog summing circuit **77**. As the volume of the customer's voice increases, the voltage on the output becomes increasingly negative.

The ambient noise control **81** as shown in FIG. **3A** comprises a variable resistance between a negative voltage supply and the input to the summing circuit **77**. The variable resistance is preset to apply a negative voltage to an input of summing circuit **77** which is sufficiently negative to prevent ambient noises of the teller location from causing the communication direction to change.

The summing circuit 77 comprises an electrical connection between the three inputs noted in FIG. 1 which is electrically connected to the input of an integrated circuit amplifier in the connection as shown in FIG. 3A. The output of the summing circuit 77 is applied to the wire 82 and comprises a negative signal which is applied to the control input of the voltage control amplifier 42. The negative signal on the output line 82 has a magnitude which is proportional to the input signal to the summing circuit 77 having the greatest negative magnitude.

The negative output signal from the summing circuit 77 is applied to the control voltage input of the voltage control amplifier 42. As the voltage output of the summing circuit 77 becomes increasingly negative, the gain of the voltage control amplifier is reduced. Consequently, the signals from the teller microphone 40 are not amplified as much by the voltage-controlled amplifier 42 when the output of summing circuit 77 becomes increasingly negative. The reduced gain will prevent the level-sensing circuit 55 from detecting a feedback signal loud enough at the teller microphone to cause the communication direction to change, thus insuring that the teller is in control of the communication direction.

The feedback controls also operate to produce an automatic volume control for the teller's voice. When the teller has switched the communication system to permit his voice to be reproduced by the two-way speaker 10, the resistor 83 is coupled between the power amplifier output 39 and the input to the automatic volume control amplifier 78. This signal is fed back to the voltage control amplifier 42 and automatically adjusts the microphone gain while the teller is speaking and is operative to automatically prevent the teller from producing a very high volume signal at the two-way speaker 10 by reducing the voltage-controlled amplifier gain 42 as the teller's voice level increases.

In a remote banking installation where more than one teller is available to assist more than one customer, a switching network is required to allow the teller to select which customer location he will assist. Such a network is shown in FIG. 4 where, for example, two teller locations and three customer locations are shown. This system includes a microphone and a speaker at each teller location and a two-way speaker at each customer location. There is an amplifier and associated control circuits 124 connected to the microphone and speaker at each teller location, each amplifier and associated controls comprising a network like that shown in FIGS. 3A and 3B. The customer connection points 125 and 126 correspond to the input line 11 and serve to carry the electrical signals going to and coming from the customer location. The microphone inputs 127 and 128 for the teller locations correspond to the teller microphone 40 connection, with the preamplifier 41 and the teller speaker connections 129 and 130 corresponding to the speaker connection to point 36.

To permit the tellers and customers to communicate with each other, a switching network 131 is provided for connecting the two-way customer speakers with the amplifier and associated control circuit 124 for the assisting teller. The switching network may take the form of a rotary stepping switch at each teller location. The teller simply sets the switch to a position corresponding to the customer location to which the teller desires to communicate and the desired customer two-way speaker is connected to the amplifier and associated controls 124. Once the communication path is established by setting the switching network as desired, the communication between the teller and the customer will proceed as described earlier for the system in FIG. 1.

A simple switching network such as described above has a serious drawback because more than one teller could attempt to communicate with the same customer location, a condition which would cause confusion. To obviate this possibility, a switching network like that in FIG. 5 may be provided which, for example, is for a system with three teller and six customer locations. The network shown in FIG. 5 is associated with the second of the three teller locations and identical networks must be provided for each of the other two teller locations. The network shown in FIG. 5 is operative to prevent more than one teller from attempting to communicate with the same customer location at the same time.

The circuitry of FIG. 5 is best described by way of example. Each teller location is provided with a plurality of customer selection switches 150, each switch comprising a plurality of single-pole, single-throw (SPST) switches ganged together and all ganged switches being simultaneously closed or opened. Each customer switch has as many SPST switches as there are teller locations in the system. Each of the switches has one contact grounded and the other contact connected to a wire labeled, for example, 2B. This labeling represents the teller No. 2 and the customer number (B). Whenever a teller attempts to communicate with a customer location, each of the ganged SPST switches associated with a given customer and a O signal will be placed on each of the connected lines, such as line 2B.

Assuming teller No. 2 wishes to communicate with customer B, teller No. 2 will close the switch 151 placing a ground or 0 level on the wire labeled 2B. The inverter 153 will produce a 1 level output when the input is a 0. The wire 2B is also connected to the NAND circuit 154 which produces a 1 level output whenever any input becomes a 0. The output of NAND 154 is one input to NAND 155 and will remain at the 1 level as long as teller No. 2 has one of his customer selector switches closed.

The select B signal appearing as a 1 level at the output of inverter 153 is applied to one input of the NAND circuit 156 which serves as a driver circuit for the relay coil K1. The select B signal is also applied to one input of NAND circuit 152. NAND circuit 152 produces a 0 output when teller No. 2 wishes to select customer B and the other two tellers are not selecting customer B, the latter condition is indicated by 1 level signals on lines 160 and 161 which connect to the B customer switches at the other two teller locations.

The output of NAND 152 is applied to one input of NAND 157 which produces a 1 level output whenever any of the inputs is a 0. The output of NAND 157 is applied to an input of NAND 155 which combines with the 1 level signal from NAND 154 to produce a 0 level output as long as teller No. 2 has switch 151 closed. This 0 level signal from NAND 155 is applied to one of the inputs to NAND 157 and is operative to form a latch circuit which will remain set until switch is opened. This latch circuit is necessary because the 0 signal at the output of NAND 152 would become a 1

level signal if any of the other tellers subsequently attempts to communicate with customer B, causing the NAND 152 output to switch to the 1 level.

The output of NAND 157 is also applied to a second input to NAND 156. When both inputs to NAND circuit 156 are a 1, the output will be the 0 which causes current to flow through the K1 relay coil. Activating the K1 relay causes the K1 relay contact to close which then connects the amplifier and associated controls for teller No. 2 with the two-way speaker at the customer B location. Consequently, when a teller attempts to assist a given customer location, the switching network first determines whether that customer location is being assisted by another teller. If this condition does not exist, then the teller selecting the specific customer location is connected to the selected customer two-way speaker. Communication between the teller and the selected customer location will continue until the teller changes his selector switch. For example, when teller No. 2 opens switch 151, a 1 level signal will appear on line 149. Consequently, the output of NAND circuit 154 will become a 0 which operates to reset the latch circuit comprised of NAND circuit 155 and NAND circuit 157. The 1 level signal on line 149 is applied to inverter circuit 153 and produces a 0 level signal at the output. This 0 level signal is operative to produce a 1 level signal output for NAND circuit 156 which causes the K1 relay to open, thus disconnecting teller No. 2 from the customer B two-way speaker. While the above description has described the teller No. 2 selection of customer location B, the circuit operates in an identical manner when teller No. 2 attempts to select other customer locations. Similarly, the identical networks of FIG. 5 for the other teller locations function as described for the teller No. 2 location and all the switching networks combined provide the desired switching system for connecting several tellers with several customer locations while preventing more than one teller from simultaneously communicating with the same customer.

While the foregoing description of the intercom system and the different switching networks has been made with particular emphasis upon the preferred embodiments therefor, it will be clear to those of skill in the art that numerous changes in form only can easily be made without departing the spirit of this invention. Specifically, the exact circuitry for the controls and the audio amplifiers may be modified according to generally known engineering techniques without changing the fundamental operation of the system. Additionally, the logical diagram of FIG. 5 which performs the desired network switching function can take on various other forms if different logical circuits are selected. Such modifications in form only can readily be made by those of skill in the art without departing from the spirit and scope of this invention as defined in the following claims.

What is claimed:

1. An intercommunication system for a remote banking installation comprising:

speech transducing and sound reproduction means located at a customer location;

a microphone and a speaker located at a teller location, said microphone and speaker being separate electrical components;

an amplifier means with an input and an output; and

a switching means responsive to teller signals at a first nonzero level output from said microphone produced by speech input to said microphone at a first acoustic loudness to establish a first switching configuration and responsive to teller signals from said microphone at a second nonzero level less than said first level produced by speech input to said microphone at a second acoustic loudness less than said first acoustic loudness to establish a second switching configuration, said second nonzero signal level being sufficient in magnitude to permit, upon amplification by said amplifier means, audible reproduction thereof by said customer sound reproduction means, said first switching configuration providing an electrical connection from the output of said amplifier to said customer sound reproduction means and an electrical connection from said microphone to the input of said amplifier to permit the teller to talk to the customer and said second switching configuration providing an electrical connection from the output of said amplifier to said teller speaker and an electrical connection from said customer speech transducing means to the input of said amplifier to permit the customer to talk to the teller, said switching means being unresponsive to a decrease in teller speech signals to a level below said first level, but in excess of said second level, to permit the teller speech signals output from the microphone to drop below said first level to a level exceeding said second level without changing said switching configuration,

said switching means being operative normally in said second switching configuration to facilitate communication from said customer to said teller without the switching of said switch means.

2. The interconnection system in claim 1 wherein said switching means includes sequential switching means for disconnecting said speaker from said output and disconnecting said speech transducing and sound reproducing means from said input followed by the delayed connection of said speech transducing and sound reproducing means to said output and the connection of said microphone to said input when the electrical configuration is switched from said second to said first configuration and for disconnecting said speech transducing and sound reproducing means from said output and disconnecting said microphone from said input followed by the delayed connection of said speech transducing and sound reproducing means to said input and the delayed connection of said speaker to said amplifier output when the electrical configuration is switched from said first to said second configuration.

3. The intercommunication system of claim 1 wherein said second level is at least 6 decibels lower than said first level.

4. The intercommunication system of claim 1 additionally including:

a microphone sensitivity control means connected between said microphone and said switching means for reducing said microphone signals when the output from said speaker increases said sensitivity control operative to prevent customer signals reproduced by said speaker from changing said switching configuration from said second to said first switching configuration.

5. The intercommunication system in claim 4 wherein said sensitivity control means additionally in-

cludes an ambient noise control means to reduce the microphone sensitivity to ambient noise levels at said microphone.

6. An intercommunication system for a remote banking installation comprising in combination:

a two-way speaker means at a customer location;

a microphone and a speaker at a teller location, said microphone and speaker being separate electrical components;

an amplifier means with an input and an output for amplifying audio signals;

a customer preamplifier including a volume control means located at the teller location, an input and an output;

a switching network means for electrically connecting in a first switching mode said two-way speaker to said customer preamplifier input, said customer preamplifier output to said amplifier input and said amplifier output to said speaker permitting a customer to speak to a teller, and for electrically connecting in a second switching mode said microphone to said amplifier input and said amplifier output to said two-way speaker permitting a teller to speak to a customer;

a teller voice amplitude-sensing means for sensing the amplitude of signals produced by said microphone and producing a first output control signal whenever the signals produced by said microphone, in response to speech input thereto at a first acoustic loudness, exceed a first predetermined nonzero level and producing a second output control signal whenever the signals produced by said microphone, in response to speech input thereto at a second acoustic loudness less than said first acoustic loudness, fall below a second predetermined nonzero level, said second predetermined nonzero level being less than said first predetermined nonzero level, but sufficient in magnitude to permit, upon amplification, audible reproduction thereof by said customer two-way speaker, and

control means responsive to said voice amplitude-sensing means to connect said switching network to its second mode in response to said first output control signal and to connect said switching network to its first mode in response to said second output control signal, permitting the teller's voice level to fall to said second predetermined nonzero level, lower than said first predetermined nonzero level, without causing said switching mode to switch from said second to said first switching mode,

said control means being operative to normally connect said switching network to its first mode to facilitate communication from said customer to said teller without switching said switch network.

7. The intercommunication system of claim 6 wherein said switching network includes a sequence switching control means producing a first switching signal for sequentially switching said speaker from said amplifier output and said customer preamplifier output from said amplifier input followed by a second switching signal to connect said two-way speaker to said amplifier output and said microphone to said amplifier input when said network switches from said first mode to said second mode and for producing a third switching signal for switching said microphone from said amplifier input and said two-way speaker from said amplifier output to said customer preamplifier input followed by a fourth switching signal switching said customer preamplifier output to said amplifier input and said amplifier output to said speaker when said network switches from its second mode to its first mode.

8. The intercommunication system in claim 6 wherein said control means includes a sequential switching network comprising means responsive to said output signal changing from said first to said second level for connecting said two-way speaker to said customer preamplifier input and disconnecting said microphone from said amplifier input followed by connecting said speaker to said amplifier output and connecting said preamplifier output to said amplifier input, said sequential switching means responsive to said output level changing from said second to said first level for disconnecting said speaker from said amplifier output and disconnecting said preamplifier output from said amplifier input followed by connecting said microphone to said amplifier input and connecting said two-way speaker to said amplifier output.

9. An intercommunication system for a remote banking installation comprising in combination:

a two-way speaker means at a customer location;

a microphone and a speaker at a teller location, said microphone and speaker being separate electrical components;

an amplifier means with an input and an output for amplifying audio signals;

a customer preamplifier including a volume control means located at the teller location, an input and an output;

a switching network means for electrically connecting in a first switching mode said two-way speaker to said customer preamplifier input, said customer preamplifier output to said amplifier input and said amplifier output to said speaker permitting a customer to speak to a teller, and for electrically connecting in a second switching mode said microphone to said amplifier input and said amplifier output to said two-way speaker permitting a teller to speak to a customer;

a teller voice amplitude-sensing means for sensing the amplitude of signals produced by said microphone and producing a first output control signal whenever the signals produced by said microphone, in response to speech input thereto at a first acoustic loudness, exceed a first predetermined nonzero level and producing a second output control signal whenever the signals produced by said microphone, in response to speech input thereto at a second acoustic loudness less than said first acoustic loudness, fall below a second predetermined nonzero level, said second predetermined nonzero level being less than said first predetermined nonzero level, but sufficient in magnitude to permit, upon amplification, audible reproduction thereof by said customer two-way speaker,

control means responsive to said voice amplitude-sensing means to connect said switching network to its second mode in response to said first output control signal and to connect said switching network to its first mode in response to said second output control signal, permitting the teller's voice level to fall to said second predetermined nonzero level, lower than said first predetermined nonzero level, without causing said switching mode to

switch from said second to said first switching mode, said control means comprising:

a delay means connected to the output of said voice amplitude-sensing means for producing at its output delayed signals from said amplitude-sensing means;

a NOR circuit having two inputs and one output, one said NOR input being connected to said voice amplitude-sensing means output and said other NOR input being connected to said delay output, said NOR output producing a first switching signal when said voice amplitude-sensing means output or said delay output is at said first level and said NOR output producing a second switching signal when said voice amplitude-sensing means output and said delay output are at said second level; and

an AND circuit with two inputs and one output, one input to said AND circuit being connected to said delay means output and said other AND input being connected to the output of said voice amplitude-sensing means, said AND circuit output producing a third switching signal when said voice amplitude-sensing means output is at its first level and said delay output is also at said first level, said AND circuit output producing a fourth switching signal when either input is at said second level of said voice amplitude-sensing means output, said first switching signal operative to disconnect said speaker from said amplifier output and to disconnect said preamplifier output from said amplifier input said second switching signal operative to connect said microphone to said amplifier input and said two-way speaker to said amplifier output, said third switching signal operative to connect said two-way speaker to said preamplifier input and disconnect said microphone from said amplifier input and said fourth switching signal operative to connect said speaker to said amplifier output and said preamplifier output to said amplifier input.

10. The intercommunication system in claim **6** additionally including:

a microphone gain control means for controlling the amplitude of the signals produced by said microphone and applied to the input of said voice amplitude-sensing means; and

means for reducing the gain of said microphone gain control means in response to increased amplitude signals at said amplifier output when said switching network is in said first mode preventing the customer's voice reproduced by said speaker from causing said switching network from changing from said first mode to said second mode.

11. The intercommunication system in claim **10** additionally including an impedance connected between said amplifier output and said gain reducing means when said switching network is in said second mode for providing an automatic volume control for the teller's communications to the customer.

12. The intercommunication system in claim **10** additionally including an adjustable microphone gain control means to set the maximum microphone gain for preventing ambient noise at the microphone from switching said switching network from said first mode to said second mode.

13. The intercommunication system of claim **6** additionally including:

a muting means for disconnecting said speaker from said amplifier output when said switching network is in said first mode, said muting means being responsive to a mute signal generated by a system associated noise producing apparatus at the customer location.

14. The intercommunication system of claim **6** additionally including:

at least one other two-way speaker located at one other customer location; and

a speaker switch means for electrically connecting only one two-way speaker to said switching network means.

15. The intercommunication system of claim **6** additionally including:

at least one other two-way speaker at another customer location;

at least one other intercommunication system as defined in claim **6**; and

a second switching network means between each said two-way speaker and each said switching network operative to connect only one two-way speaker to only one switching network, permitting each teller to communicate with one customer at a time.

16. The intercommunication system of claim **15** additionally including at least one additional two-way speaker means connected to said second switching network.

* * * * *

UNITED STATES PATENT OFFICE

CERTIFICATE OF CORRECTION

Patent No. 3,849,603 Dated November 19, 1974

Inventor(s) Steve Proios

It is certified that error appears in the above-identified patent and that said Letters Patent are hereby corrected as shown below:

Column 4, line 54, change "amplififer" to --amplifier--.

Column 5, line 18, change "input 32" to --output 32--.

Column 6, line 20, insert the word "circuit" between the words "pause" and "rectifies".

Colum 7, line 24, delete the word "charging", second occurrence.

Signed and sealed this 4th day of March 1975.

(SEAL)
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United States Patent [19]

Okubo et al.

[11] Patent Number: 5,450,494

[45] Date of Patent: Sep. 12, 1995

[54] **AUTOMATIC VOLUME CONTROLLING APPARATUS**

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[21] Appl. No.: **102,007**

[22] Filed: **Aug. 4, 1993**

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[51] **Int. Cl.**[6] **H03G 3/24**

[52] **U.S. Cl.** **381/57**; 381/108

[58] **Field of Search** 381/57, 107, 108, 109, 381/71; 379/388

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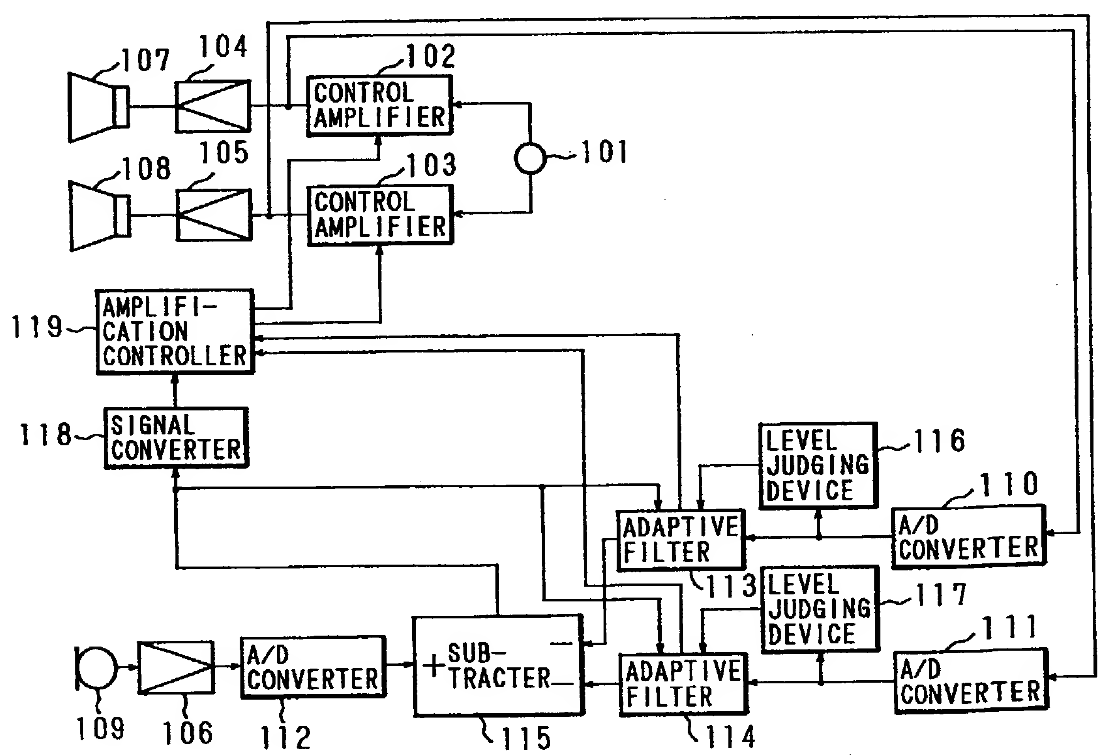
Primary Examiner—Forester W. Isen

[57] **ABSTRACT**

An apparatus for automatically controlling the sound volume of a sound producing apparatus based on ambient noise includes a microphone which detects the sound produced by the sound producing apparatus and ambient noise. Adaptive filters approximate, based on signals sent from the sound producing apparatus to speakers thereof, the characteristics of the sound components, relating to the sound produced by an individual speaker of the sound producing apparatus, which are received by the microphone. A subtractor subtracts the output of the adaptive filters from the output of the microphone to obtain a signal representing the ambient noise. The adaptive filters receive the signal representing the ambient noise and use this signal as a coefficient updating signal. A signal converter converts the signal representing ambient noise into an ambient noise level representing the volume of ambient noise. A controller then controls the gain by which amplifiers amplify signals sent from the sound producing apparatus to speakers thereof based on the ambient noise level.

8 Claims, 20 Drawing Sheets



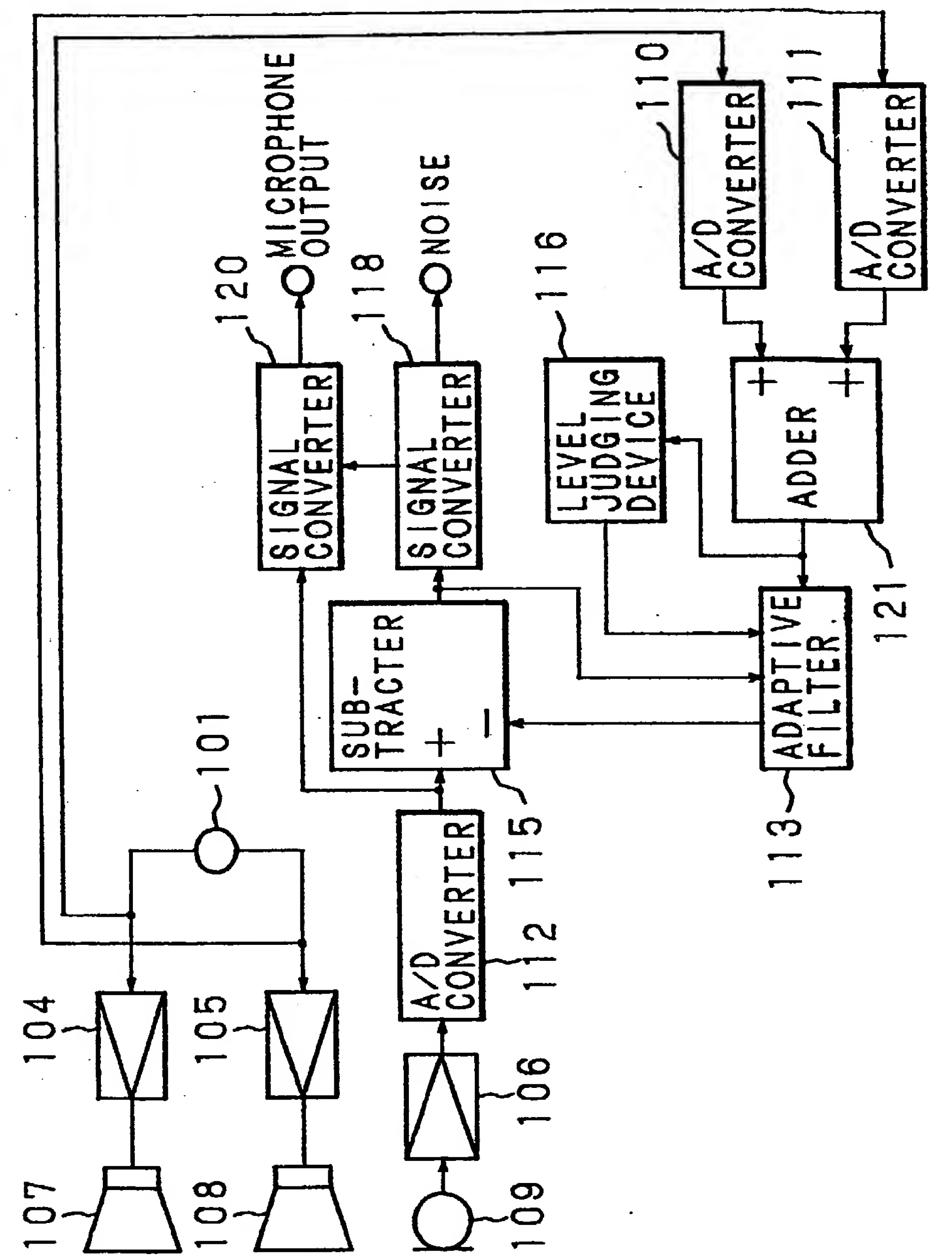


Fig. 1
Prior Art

Fig. 2
Prior Art

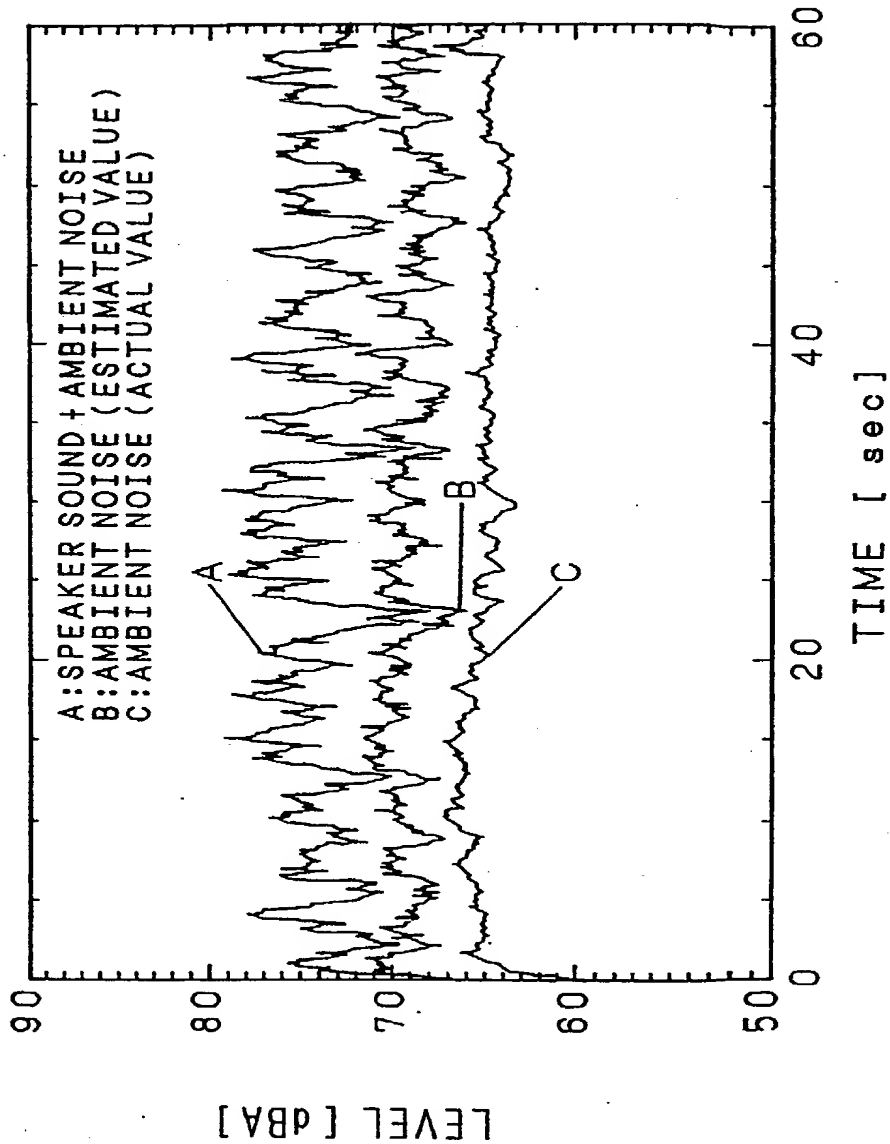


Fig. 3
Prior Art

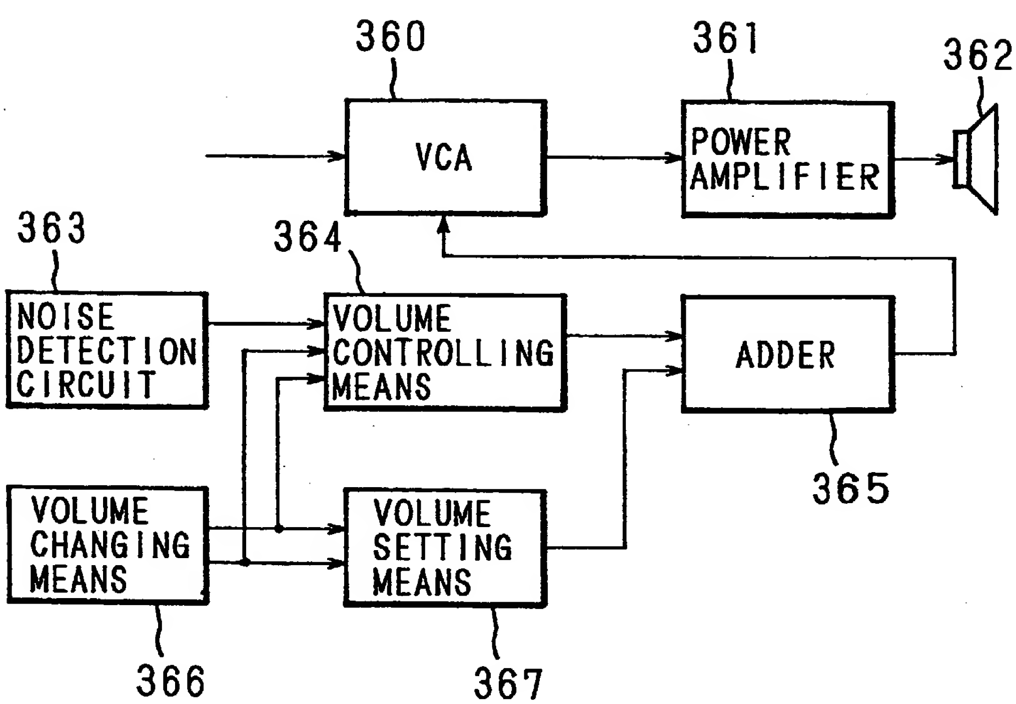
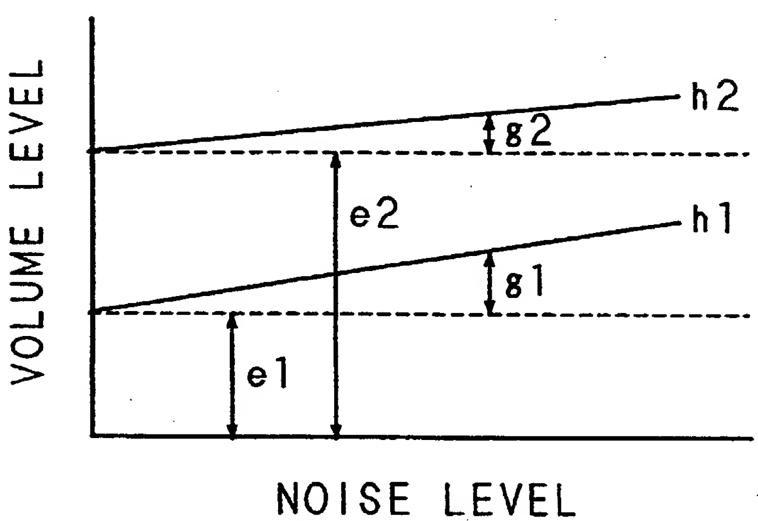


Fig. 4
Prior Art



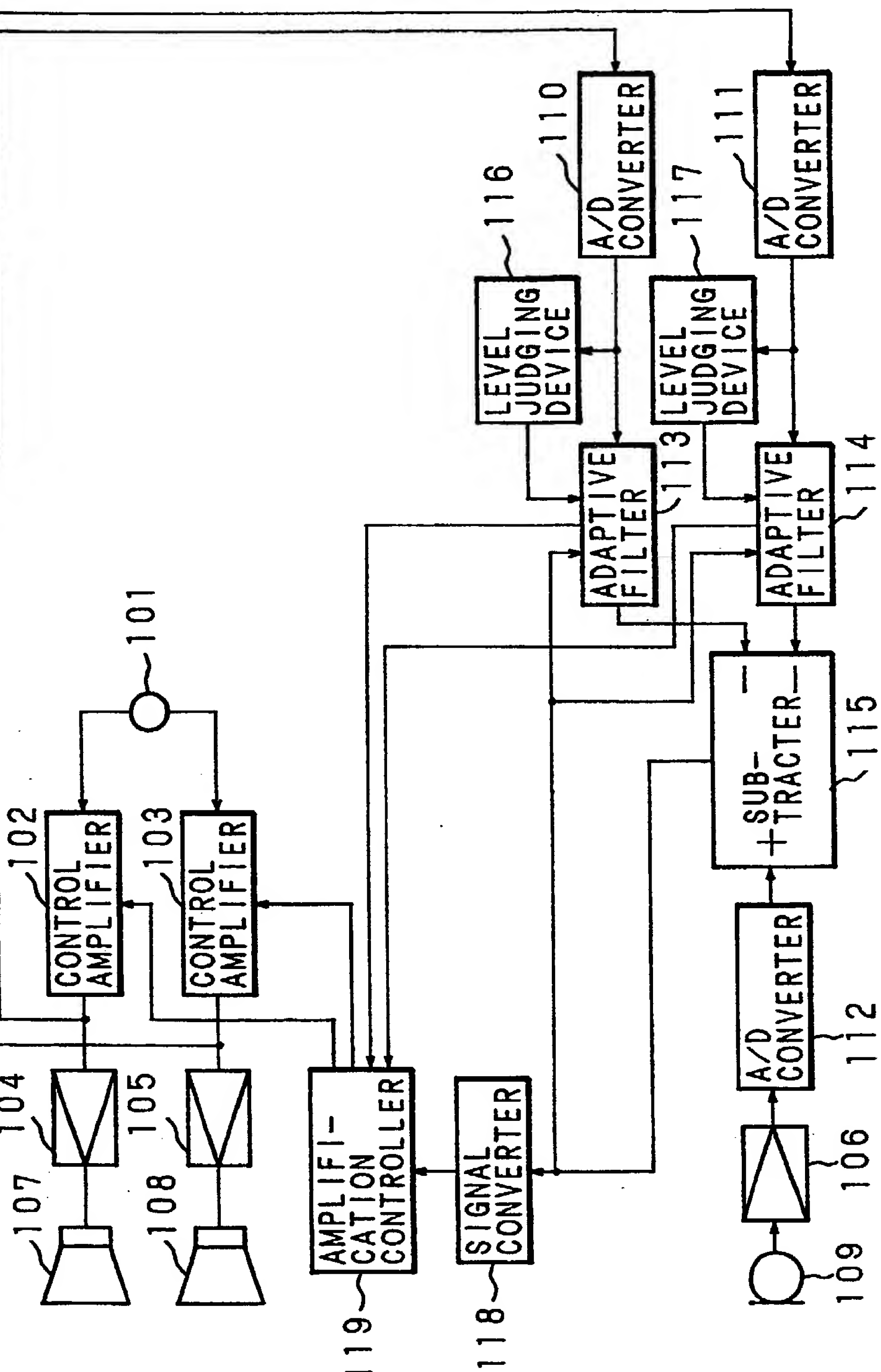

101
CONTROL AMPLIFIER
102
CONTROL AMPLIFIER
103
104
105
107
108
AMPLIFI-CATION CONTROLLER
119
SIGNAL CONVERTER
118
A/D CONVERTER
110
A/D CONVERTER
111
LEVEL JUDGING DEVICE
116
LEVEL JUDGING DEVICE
117
ADAPTIVE FILTER
113
ADAPTIVE FILTER
114
SUB-TRACTER
115
A/D CONVERTER
112
106
109

Fig. 5

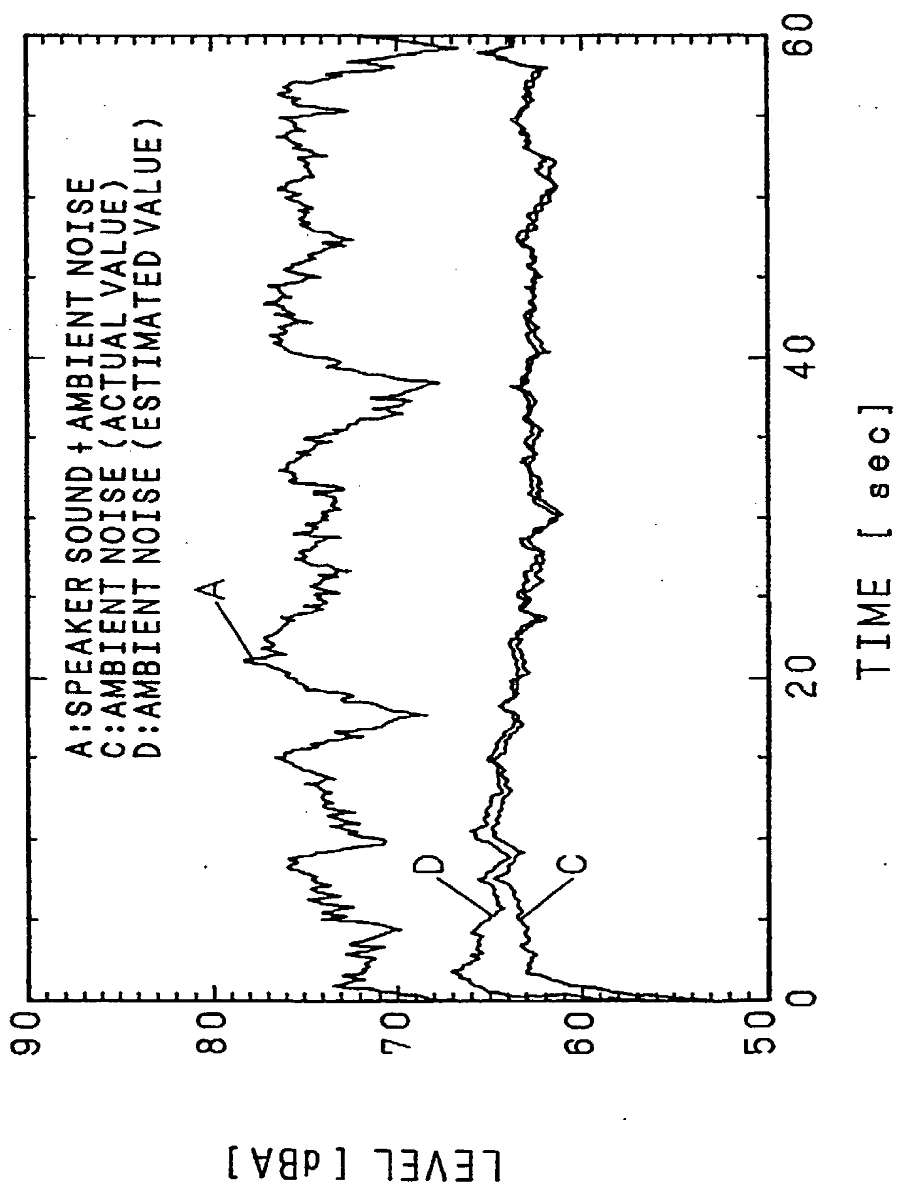

A:SPEAKER SOUND + AMBIENT NOISE
C:AMBIENT NOISE (ACTUAL VALUE)
D:AMBIENT NOISE (ESTIMATED VALUE)
A
D
C
90
80
70
60
50
0
20
40
60
LEVEL [dBA]
TIME [sec]

Fig. 6

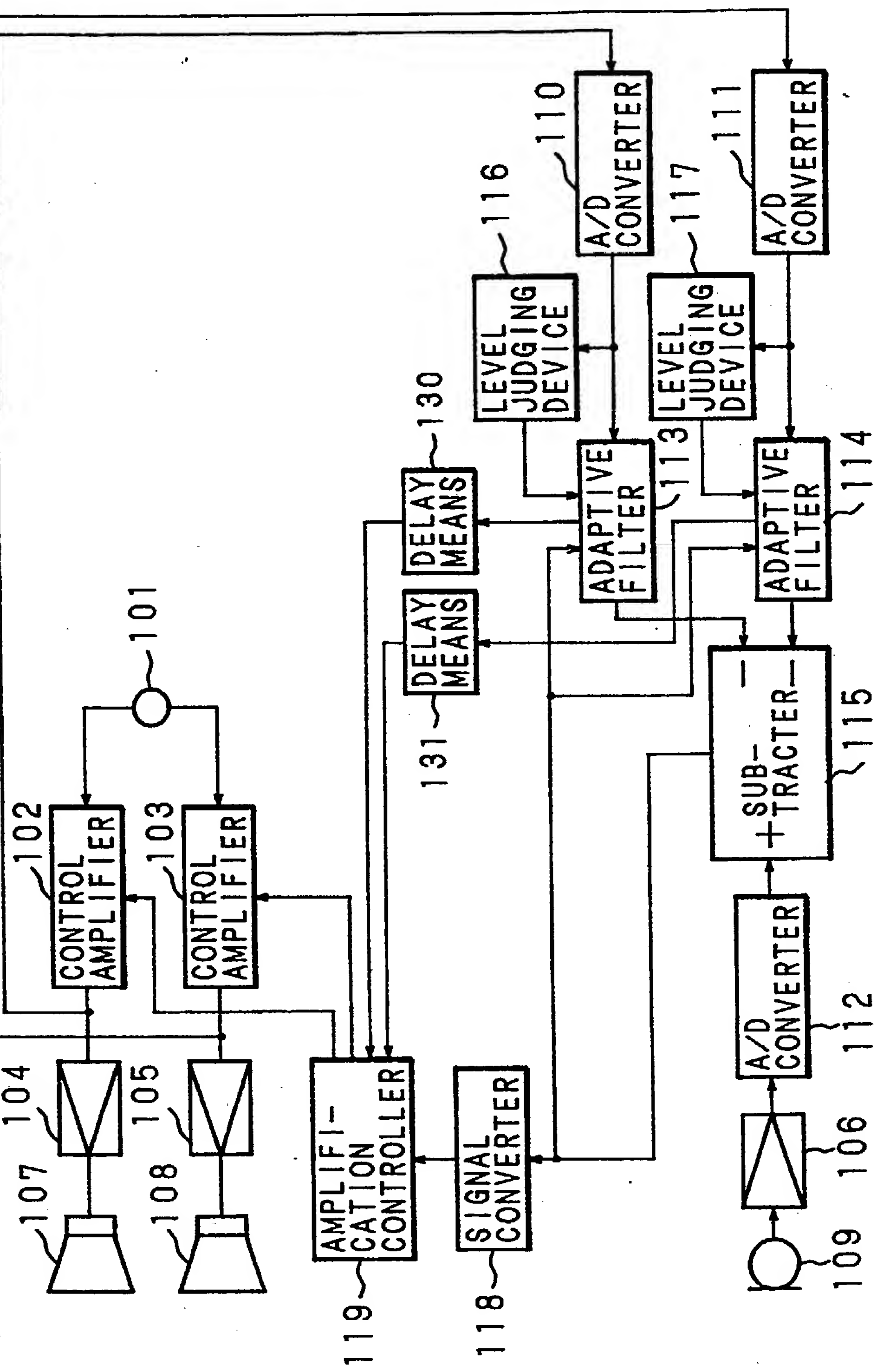
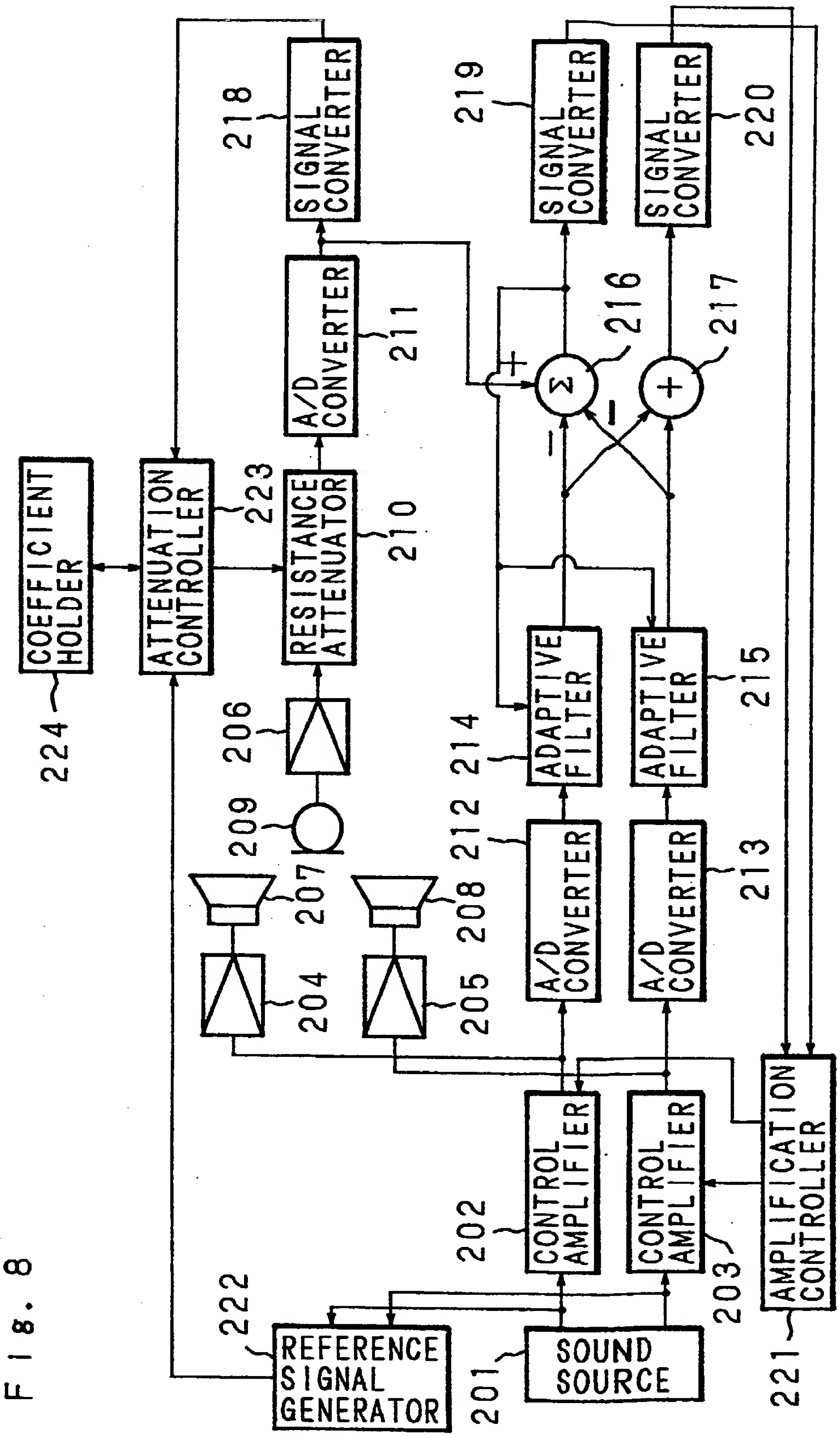

101
102
CONTROL AMPLIFIER
103
CONTROL AMPLIFIER
104
105
106
107
108
109
110
A/D CONVERTER
111
A/D CONVERTER
112
A/D CONVERTER
113
ADAPTIVE FILTER
114
ADAPTIVE FILTER
115
+ SUB- TRACTER −
−
116
LEVEL JUDGING DEVICE
117
LEVEL JUDGING DEVICE
118
SIGNAL CONVERTER
119
AMPLIFI- CATION CONTROLLER
130
DELAY MEANS
131
DELAY MEANS

Fig. 7



COEFFICIENT HOLDER
224
ATTENUATION CONTROLLER
223
RESISTANCE ATTENUATOR
210
A/D CONVERTER
211
SIGNAL CONVERTER
218
SIGNAL CONVERTER
219
SIGNAL CONVERTER
220
216
217
+
−
ADAPTIVE FILTER
214
ADAPTIVE FILTER
215
A/D CONVERTER
212
A/D CONVERTER
213
206
209
207
208
204
205
CONTROL AMPLIFIER
202
CONTROL AMPLIFIER
203
AMPLIFICATION CONTROLLER
221
REFERENCE SIGNAL GENERATOR
222
SOUND SOURCE
201

FIG. 8

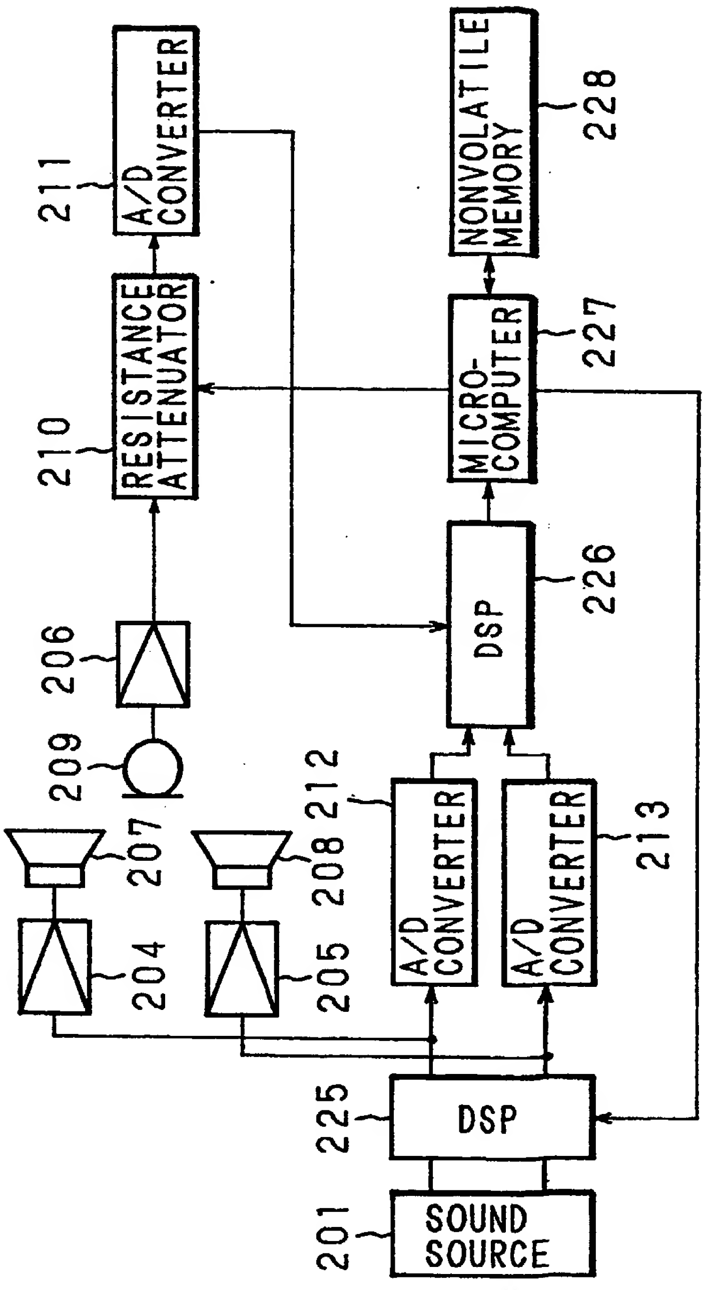

A/D CONVERTER
211
RESISTANCE ATTENUATOR
210
206
209
207
208
204
205
NONVOLATILE MEMORY
228
MICRO-COMPUTER
227
DSP
226
212
A/D CONVERTER
A/D CONVERTER
213
DSP
225
SOUND SOURCE
201

Fig. 9

Fig. 10

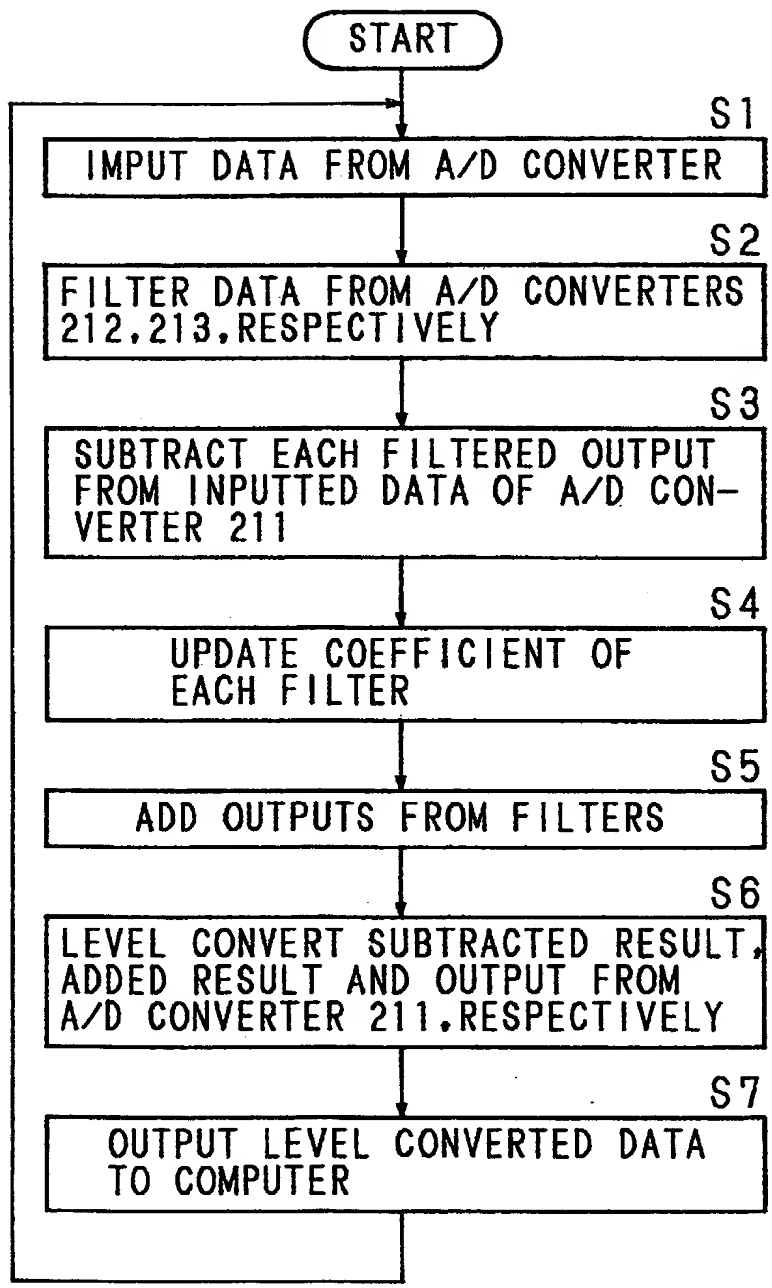

START
S1
IMPUT DATA FROM A/D CONVERTER
S2
FILTER DATA FROM A/D CONVERTERS 212, 213, RESPECTIVELY
S3
SUBTRACT EACH FILTERED OUTPUT FROM INPUTTED DATA OF A/D CONVERTER 211
S4
UPDATE COEFFICIENT OF EACH FILTER
S5
ADD OUTPUTS FROM FILTERS
S6
LEVEL CONVERT SUBTRACTED RESULT, ADDED RESULT AND OUTPUT FROM A/D CONVERTER 211, RESPECTIVELY
S7
OUTPUT LEVEL CONVERTED DATA TO COMPUTER

Fig. 11

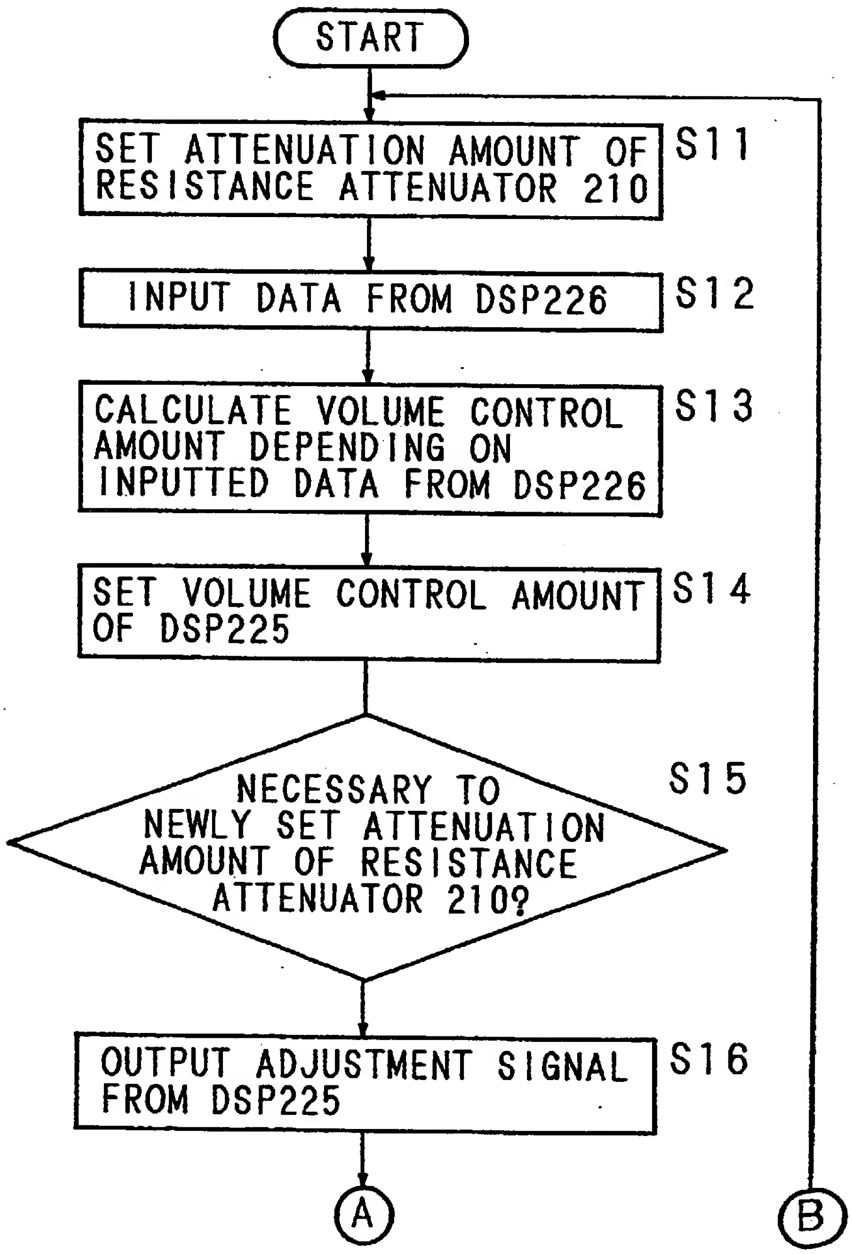
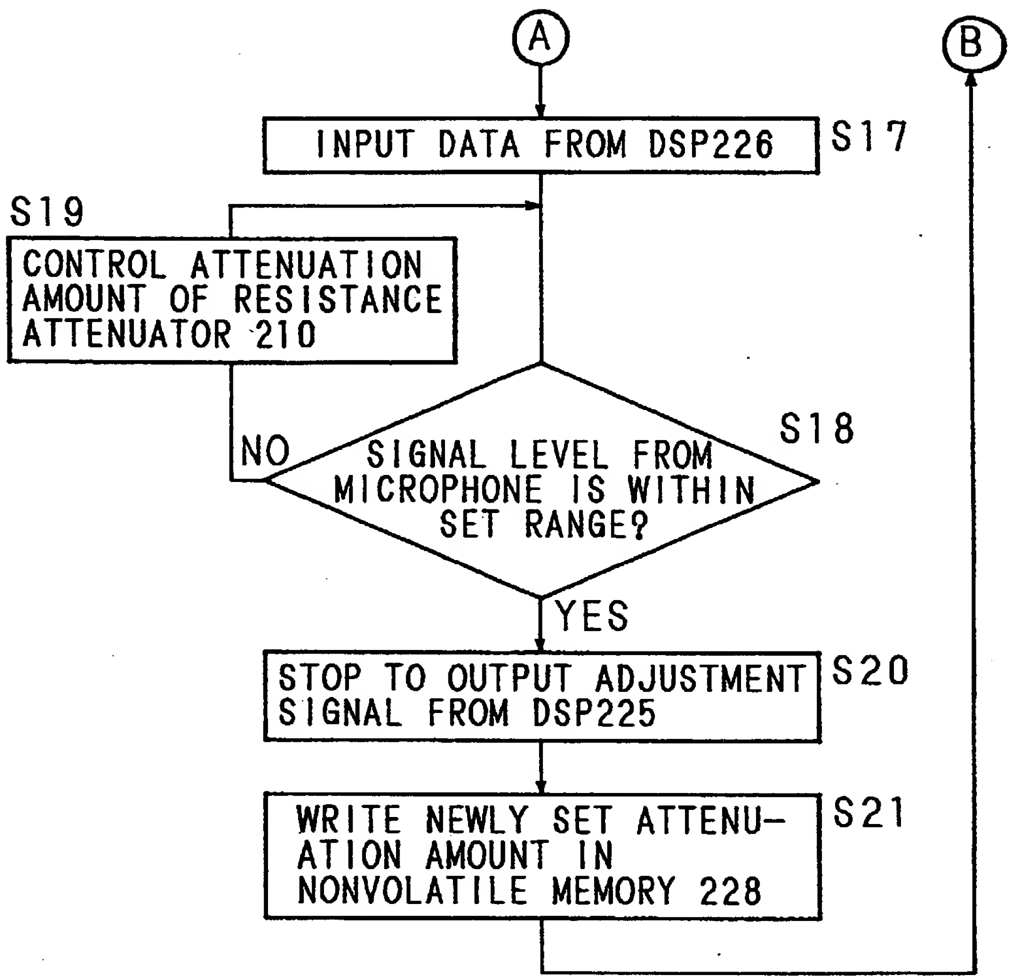
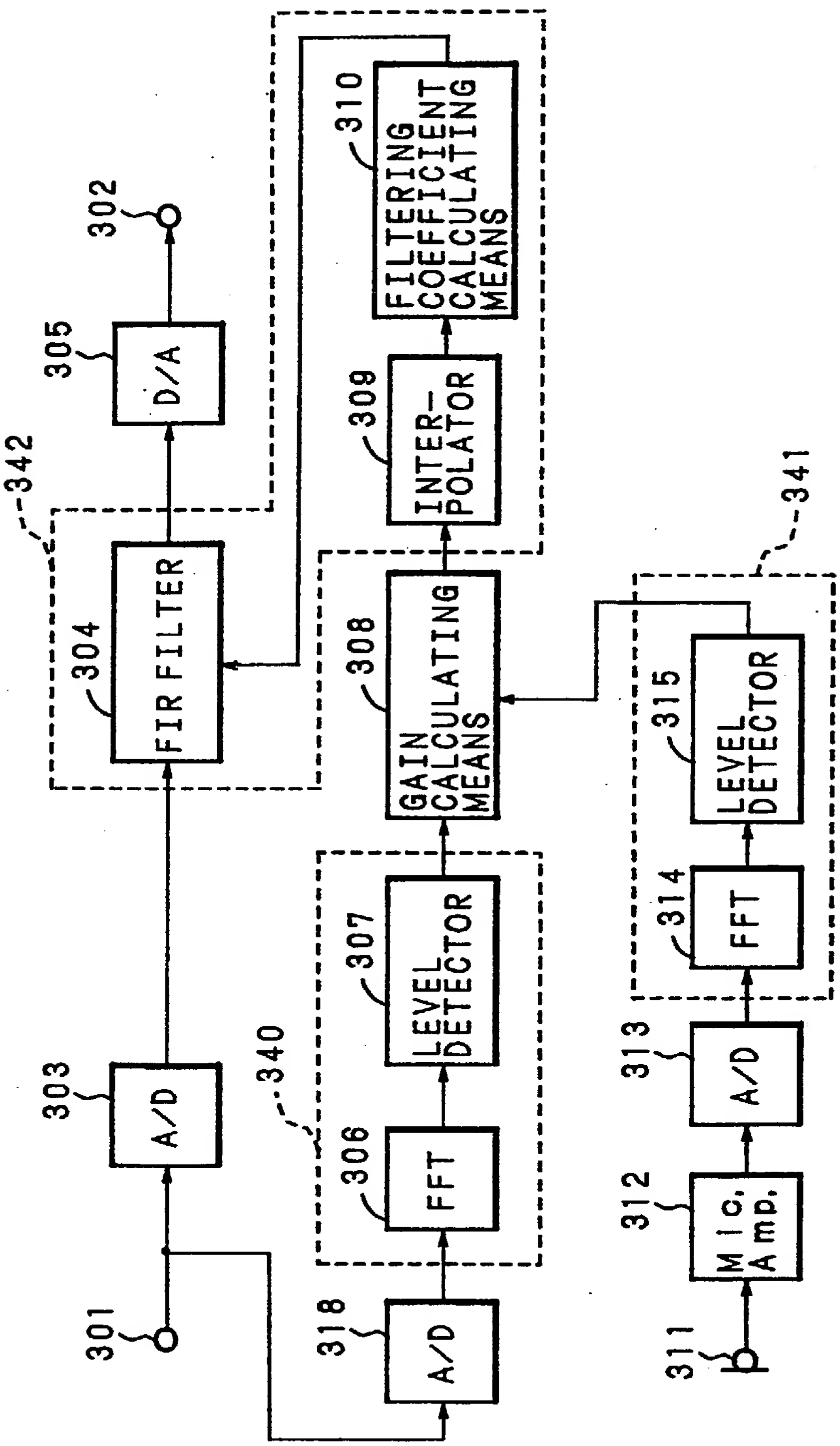

START
SET ATTENUATION AMOUNT OF RESISTANCE ATTENUATOR 210
S11
INPUT DATA FROM DSP226
S12
CALCULATE VOLUME CONTROL AMOUNT DEPENDING ON INPUTTED DATA FROM DSP226
S13
SET VOLUME CONTROL AMOUNT OF DSP225
S14
NECESSARY TO NEWLY SET ATTENUATION AMOUNT OF RESISTANCE ATTENUATOR 210?
S15
OUTPUT ADJUSTMENT SIGNAL FROM DSP225
S16
A
B

Fig. 12

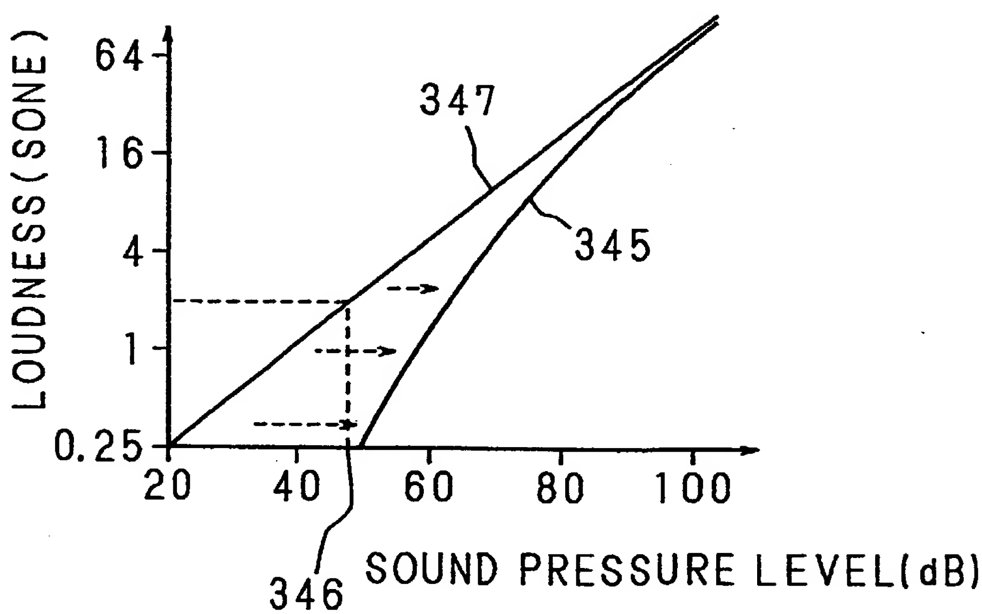

A
B
INPUT DATA FROM DSP226
S17
S19
CONTROL ATTENUATION AMOUNT OF RESISTANCE ATTENUATOR 210
NO
SIGNAL LEVEL FROM MICROPHONE IS WITHIN SET RANGE?
S18
YES
STOP TO OUTPUT ADJUSTMENT SIGNAL FROM DSP225
S20
WRITE NEWLY SET ATTENUATION AMOUNT IN NONVOLATILE MEMORY 228
S21

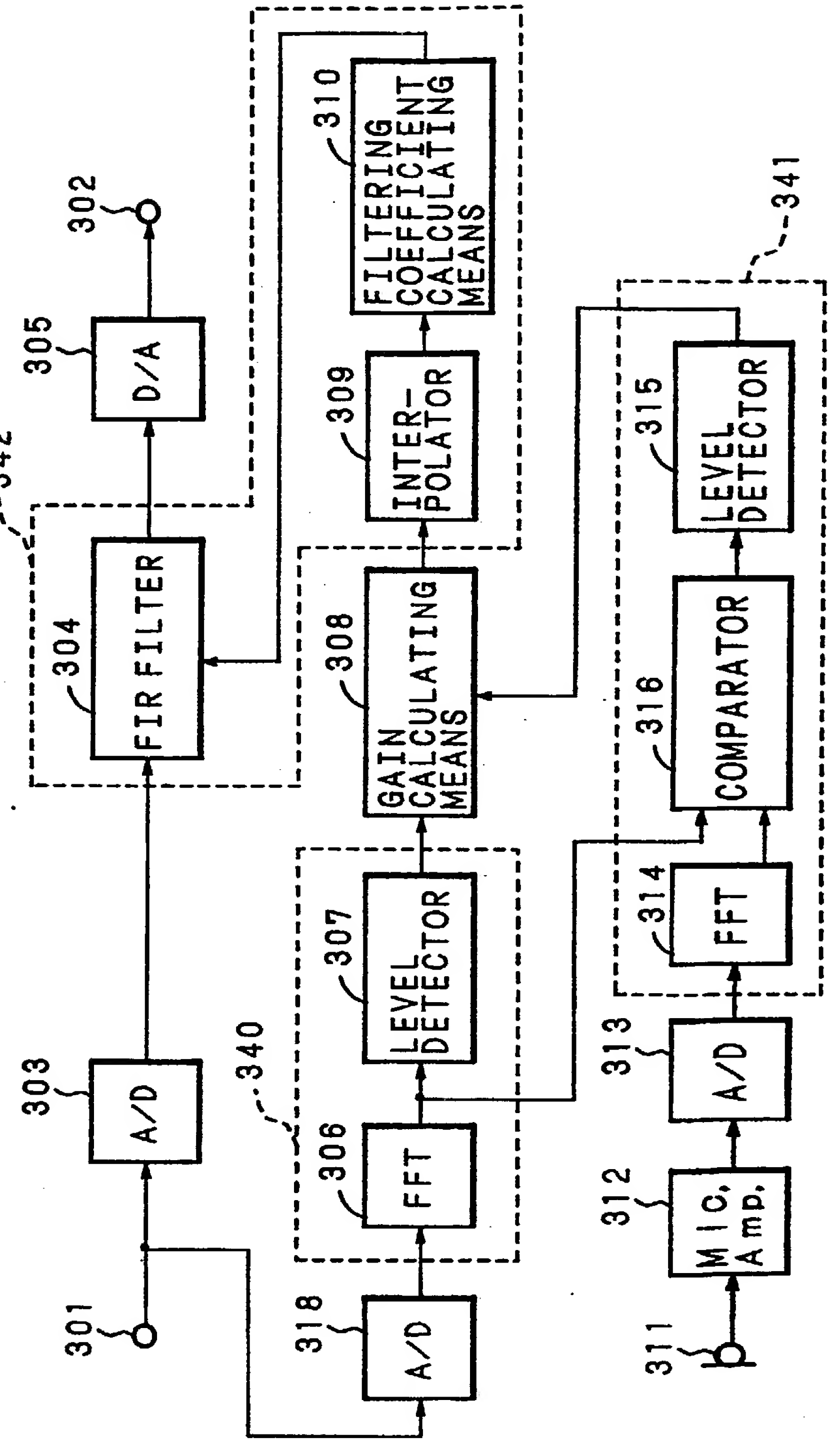


301
302
303
A/D
304
FIR FILTER
305
D/A
306
FFT
307
LEVEL DETECTOR
308
GAIN CALCULATING MEANS
309
INTER-POLATOR
310
FILTERING COEFFICIENT CALCULATING MEANS
311
312
Mic. Amp.
313
A/D
314
FFT
315
LEVEL DETECTOR
318
A/D
340
341
342

Fig. 13

Fig. 14


LOUDNESS(SONE)
64
16
4
1
0.25
347
345
20
40
60
80
100
346
SOUND PRESSURE LEVEL(dB)



301
302
303
A/D
304
FIR FILTER
305
D/A
306
FFT
307
LEVEL DETECTOR
308
GAIN CALCULATING MEANS
309
INTER-POLATOR
310
FILTERING COEFFICIENT CALCULATING MEANS
311
312
Mic. Amp.
313
A/D
314
FFT
315
LEVEL DETECTOR
316
COMPARATOR
318
A/D
340
341
342

Fig. 15

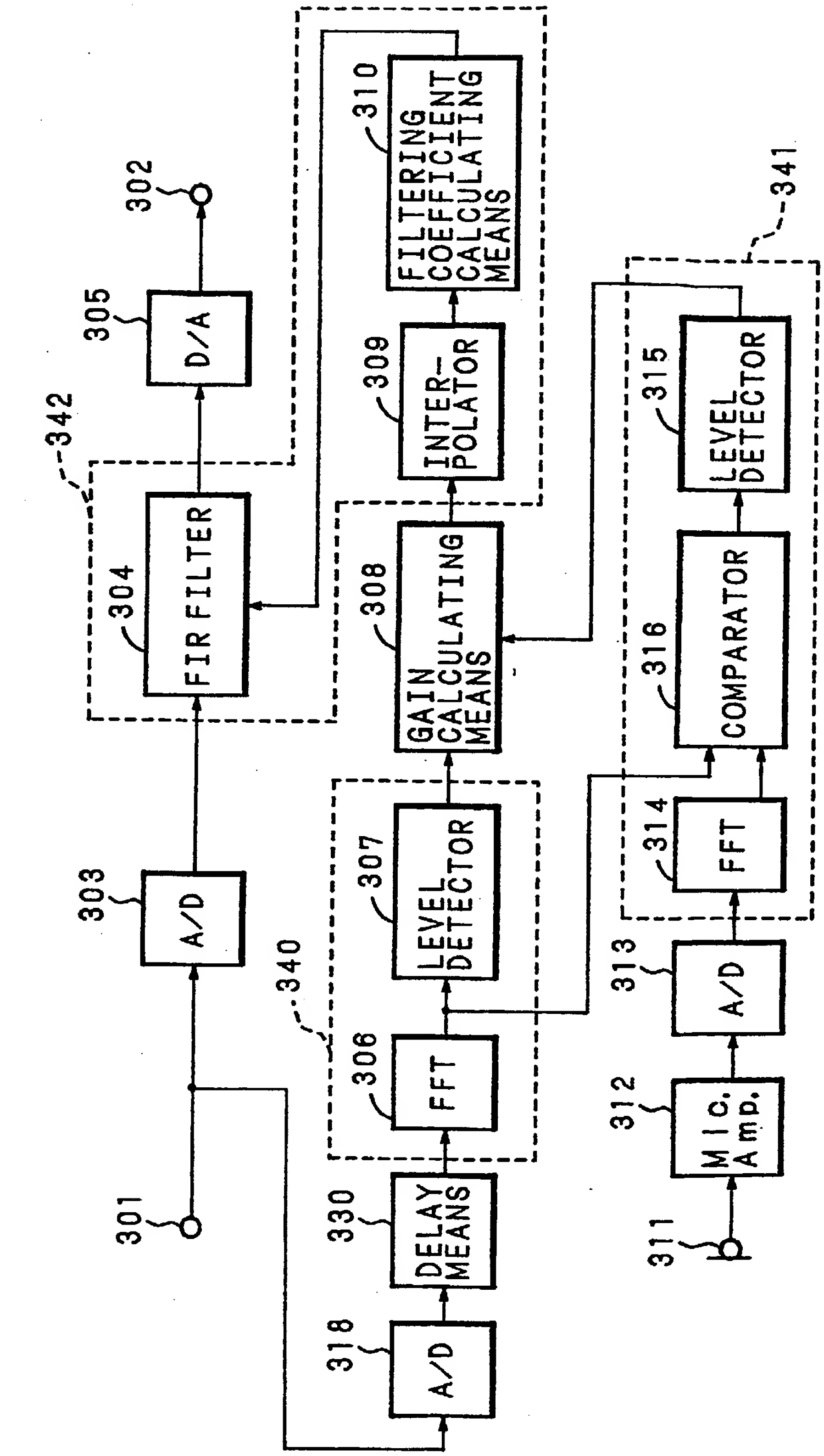

302
305
D/A
342
304
FIR FILTER
303
A/D
301
310
FILTERING COEFFICIENT CALCULATING MEANS
309
INTER-POLATOR
308
GAIN CALCULATING MEANS
307
LEVEL DETECTOR
340
306
FFT
330
DELAY MEANS
318
A/D
341
315
LEVEL DETECTOR
316
COMPARATOR
314
FFT
313
A/D
312
Mic. Amp.
311

Fig. 16

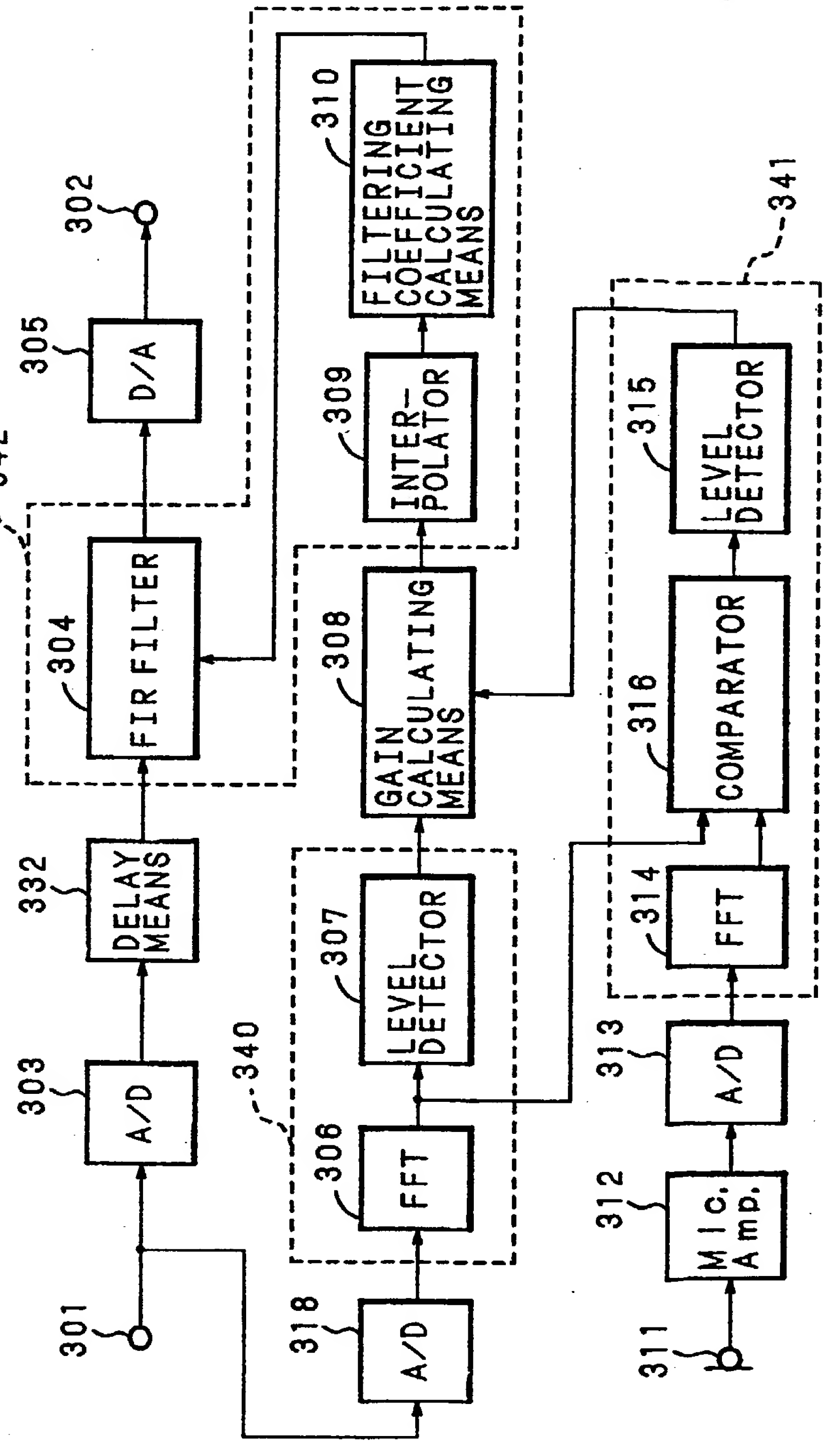

301
302
303
A/D
332
DELAY MEANS
304
FIR FILTER
305
D/A
342
318
A/D
340
306
FFT
307
LEVEL DETECTOR
308
GAIN CALCULATING MEANS
309
INTER-POLATOR
310
FILTERING COEFFICIENT CALCULATING MEANS
311
312
Mic. Amp.
313
A/D
314
FFT
316
COMPARATOR
315
LEVEL DETECTOR
341

Fig. 17

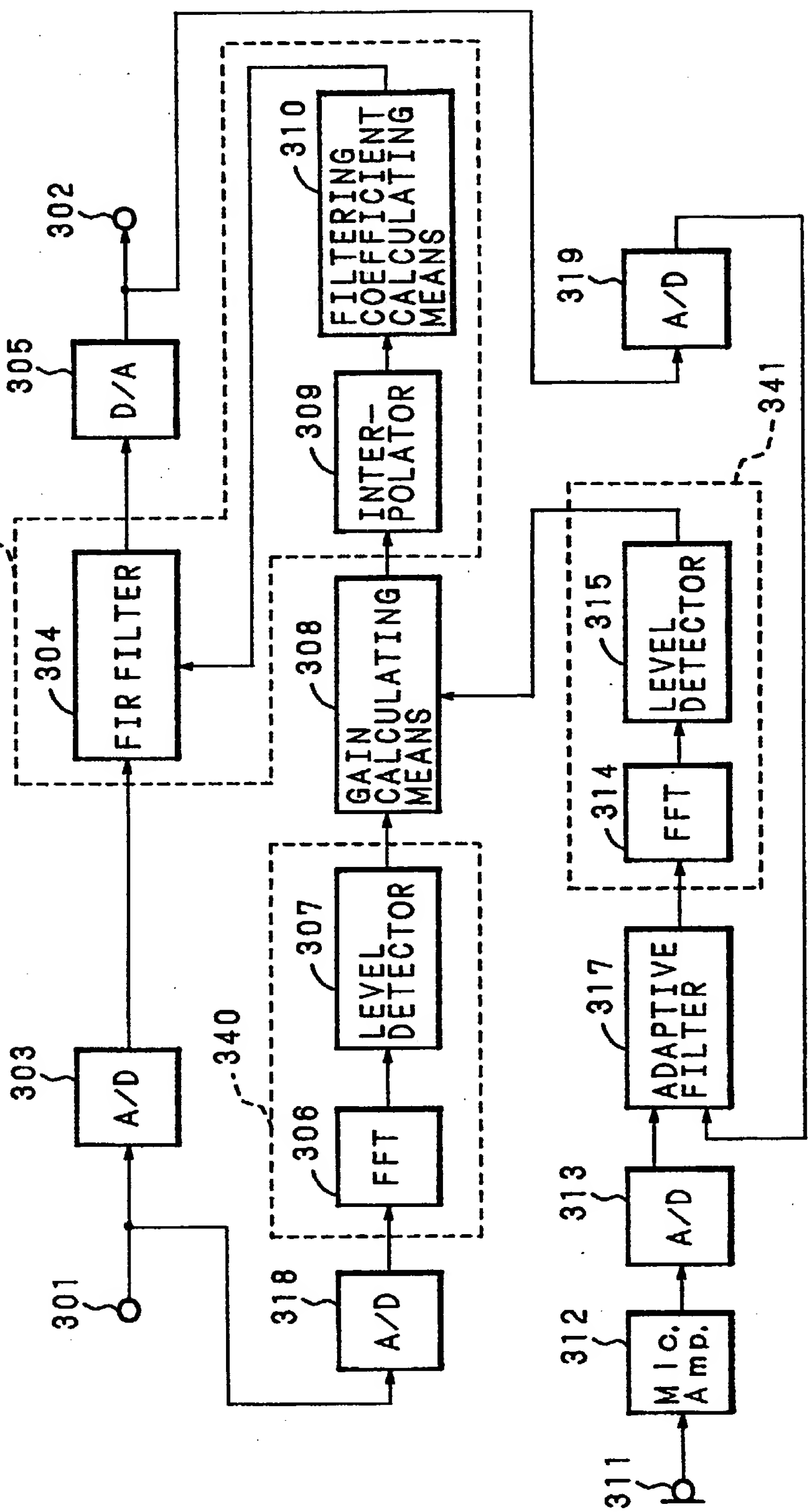
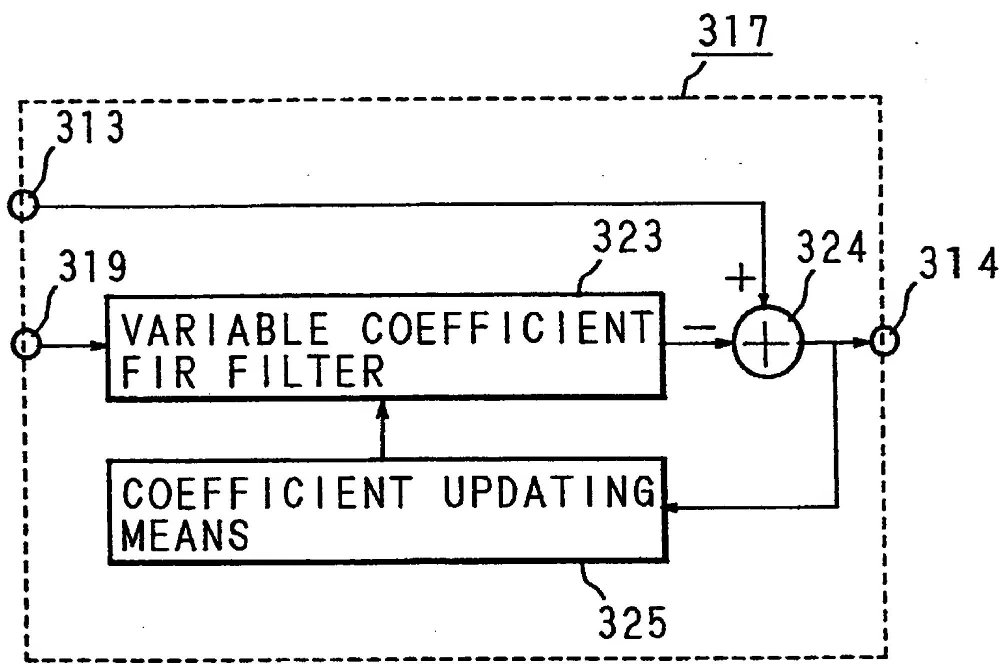

301
302
303
A/D
304
FIR FILTER
305
D/A
306
FFT
307
LEVEL DETECTOR
308
GAIN CALCULATING MEANS
309
INTER-POLATOR
310
FILTERING COEFFICIENT CALCULATING MEANS
311
312
MIc. Amp.
313
A/D
314
FFT
315
LEVEL DETECTOR
317
ADAPTIVE FILTER
318
A/D
319
A/D
340
341
342

Fig. 18

Fig. 19


317
313
319
323
324
314
+
−
VARIABLE COEFFICIENT FIR FILTER
COEFFICIENT UPDATING MEANS
325

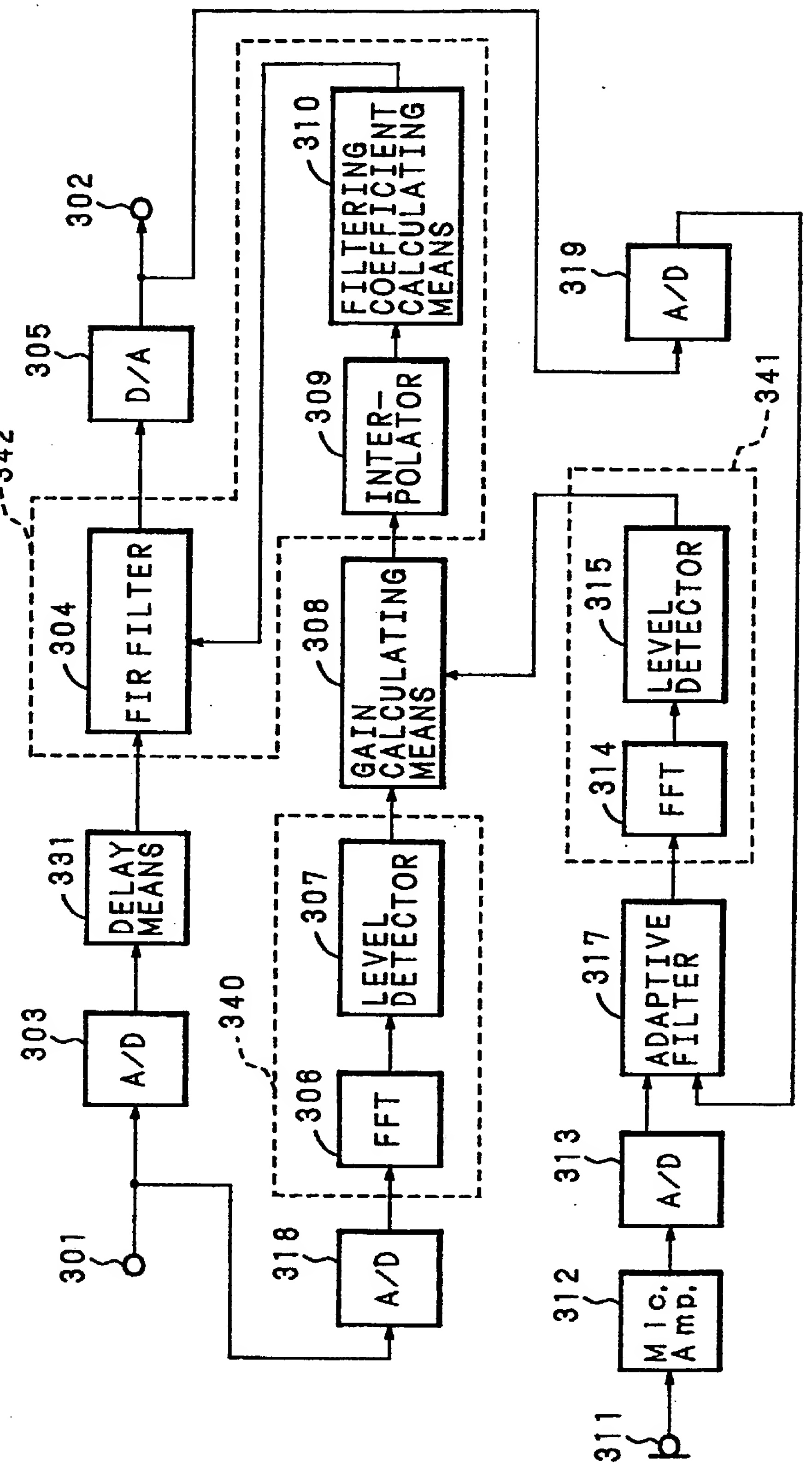

301
302
303
A/D
331
DELAY MEANS
304
FIR FILTER
342
305
D/A
318
A/D
340
306
FFT
307
LEVEL DETECTOR
308
GAIN CALCULATING MEANS
309
INTER-POLATOR
310
FILTERING COEFFICIENT CALCULATING MEANS
311
312
Mic. Amp.
313
A/D
317
ADAPTIVE FILTER
314
FFT
315
LEVEL DETECTOR
341
319
A/D

Fig. 20

AUTOMATIC VOLUME CONTROLLING APPARATUS

BACKGROUND OF THE INVENTION

1. Field of the Invention

The present invention relates to an automatic volume control for a sound reproducing apparatus used in noisy environments.

2. Description of the Related Art

In an apparatuses possessing sound output means, such as a television set, radio or a tape recorder, it has been attempted to automatically control the volume depending on the level changes in the ambient noise. As for the sound reproducing apparatus used in noisy environments, such as a car radio or a car stereo equipment, it has been attempted to automatically raise the reproducing volume depending on the level of the ambient noise so as to maintain a favorable listening condition even in a noisy situation.

Typical apparatuses are as follows: (a) an apparatus in which a mixed sound of an ambient noise with a speaker sound is detected by a microphone, and the level difference in the detected signal from the signal out of a sound signal output circuit is calculated to detect the level of the ambient noise, thereby to control the gain of the sound signal output circuit; (b) an apparatus in which the ambient noise is detected only when no sound is produced from the loudspeaker so as to obtain a sound level corresponding to the ambient noise thereby to control the volume by maintaining the same level when the loudspeaker produces sound; (c) an apparatus where an ambient noise is detected by picking up a mixed sound of the ambient noise with a speaker sound by a microphone, produces a signal imitating the speaker sound from the signal outputted from the sound signal output circuit by using an adaptive filter, then subtracting the adaptive filter output signal from the microphone output signal, thereby to control the volume depending on the ambient noise level.

As for the apparatus (a), adequate control by detecting the level of the ambient noise signal component separately from the mixed signal of the ambient noise with speaker sound is difficult because the speaker sound component detected by the microphone is greatly different from the original electric signal due to phase delay of the speaker sound, frequency response of loudspeaker, and room reverberation characteristic.

As for the apparatus (b), it is far from practical. When used in a place where the ambient noise changes suddenly or periodically while the loudspeaker is producing sound, and in particular if the sound is continuous such as with music, the apparatus (b) is almost impossible to control because the loudspeaker rarely stops producing sound.

As for the apparatus (c), it functions normally when the sound source is monaural, but has difficulty detecting the ambient noise in case of stereo. Since the sum of the right and left signals is used as the input to the adaptive filter, the apparatus (c) is affected by the positions of the microphone and the loudspeakers, or shape of the room.

FIG. 1 is a block circuit diagram showing an example of a conventional automatic volume controlling apparatus of the (c) type proposed by the present inventors in the Japanese Patent Application Laid-Open No. Hei. 5-259779 (1993). In the figure, numeral **101** denotes a sound source, **104**, **105**, **106** denote amplifiers, **107**, **108** denote loudspeakers, **109** denotes a microphone, **110**, **111**, **112** denote A/D converters, **113** denotes an adaptive filter, **115** denotes a subtracter, **116** denotes a level judging device for level converting the A/D converted signal and comparing the level with the reference levels **118**, **120** denote signal converters for level converting the outputs of the subtracter **115** and A/D converter **112**, respectively, and **121** denotes an adder for summing the outputs of the A/D converters **110**, **111**.

In this apparatus, by using the adaptive filter **113**, an imitative loudspeaker signal added with a characteristic closely approximating a signal on route from the loudspeakers **107**, **108** to the microphone **109** is canceled from the output of the microphone **109**. The level of the ambient noise is obtained as the output of the signal converter **120** when a signal is not produced from the sound source **101**. When a signal is produced from the sound source **101**, the estimated value of the ambient noise level and the level of the mixed sound of the speaker sound with ambient noise are obtained as the outputs of the signal converters **118**, **120**, respectively.

FIG. 2 is a graph showing an example of the result obtained in the apparatus shown in FIG. 1, in which A indicates the level of the mixed sound of the ambient noise with speaker sound obtained by the microphone, B indicates the level of the ambient noise detected by the conventional apparatus, and C indicates the level of the ambient noise obtained by the microphone when no sound is produced by the loudspeaker. Herein, the ambient noise is imitatively reproduced by the loudspeaker and added to the speaker sound.

As clear from FIG. 2, there is a slight difference between the level B of the detected ambient noise and the level C of the ambient noise obtained when the speaker sound is not produced, so that the ambient noise is not detected correctly.

Moreover, in this prior art, although the speaker sound level can be controlled depending on the ambient noise, the volume of the speaker sound is not considered. In this kind of the automatic volume controlling apparatus, the volume is changed relative to the ambient noise level to suit the human auditory characteristic. The volume amplification degree is changed according to the level of the reproduced sound of the sound producing apparatus. That is, the higher the level of the reproduced sound from the sound producing apparatus, the smaller the volume should be amplified in response to the same ambient noise level.

One proposal is to use an automatic volume controlling apparatus capable of varying the amplification degree relative to the noise level signal depending on the volume level set by a volume changing means. FIG. 3 is a block diagram showing a conventional automatic volume controlling apparatus disclosed in the Japanese Patent Publication No. Hei. 3-13762 (1991). In the diagram, numeral **360** denotes a voltage controlled amplifier (VCA), **361** denotes a power amplifier, **362** denotes a loudspeaker, **363** denotes a noise detection circuit, **364** denotes volume controlling means, **365** denotes an adder, **366** denotes volume changing means, and **367** denotes volume setting means.

In this constitution, a sound signal is passed through the VCA **360** and power amplifier **361** and is reproduced by the loudspeaker **362**. The amplification rate of the VCA **360** is controlled by the output of the adder **365**. To the adder **365** are inputted the noise level signal detected by the noise detection circuit **363** via the level

controlled by the volume controlling means 364, and the output from the volume setting means 376 for producing a volume level signal corresponding to the desired volume of the user on receiving a volume increase or decrease signal from the volume changing means 366. In this constitution, the volume is controlled in accordance with the noise detected by the noise detection circuit 363. The volume controlling means 364 changes the amplification degree against the noise level signal corresponding to the set volume at the time of receiving the output from the volume changing means 366. More specifically, as shown by a symbol h2 in FIG. 4, when the sound level (e2) is high, the amplification degree (g2) against the noise level is lowered to reduce the volume increase against the noise increase. As shown by a symbol h1, by contrast, when the sound level (e1) is low, the amplification degree (g1) against the noise level is raised to increase the volume increase against the noise increase, so that the volume control against the ambient noise more highly suits the auditory sense.

However, the speaker sound is actually heard, so that when the volume amplification degree is controlled depending on the volume set by the volume changing means 366, or when the volume amplification degree is controlled depending on the output from the volume controlling means 364, it is further necessary to control the volume amplification degree depending on the performance of the loudspeaker to be used. Besides, since the gain of the sound signal against the noise level is determined by representing the sound signal level with the set volume, it is impossible to avoid shortage or excess of auditory volume due to a difference in recording levels of music sources, or level changes in phrases of music.

Moreover, in the automatic volume controlling apparatus as shown in FIG. 1, the signal from the microphone 109 and the input signal to the adaptive filter 113 are usually digital signals from the A/D converter 112, but at this time since the input from the microphone to the A/D converter 112 must be within a specific range, the microphone output level must be controlled every time to the optimum value depending on the installed positions of the loudspeakers 107, 108 and microphone 109.

Incidentally, the sense of shortage of the volume due to the noise is mainly caused by masking of the sound signal by the noise. The sound signal component to be masked is larger in the frequency range of a larger noise component, and smaller in the frequency range of a smaller noise component. For example, the automobile running noise has a larger component at lower frequencies and a smaller component at higher frequencies.

Therefore, if the reproduced volume is uniformly controlled without regard to the frequency of the sound signal, as in the conventional apparatus, the effect is insufficient in the frequency range with larger noise components, but is excessive in the frequency range with smaller noise components. When such mere gain increase is done for the automobile running noise, sensation of the volume shortage in the low frequency range and sensation of the volume excess in the high frequency range are unavoidable.

As a solution, a frequency characteristic can be compensated by boosting the low frequency range of the sound signal or attenuating the high frequency range. This solution assumes the spectrum of the noise, and it may not be a sufficient solution if the noise spectrum varies.

SUMMARY OF THE INVENTION

The invention is devised to solve the above mentioned problems and it is a first object of the invention to provide an automatic volume controlling apparatus capable of automatically controlling the output volume of the loudspeaker depending on the ambient noise level detected with high precision.

It is a second object of the invention to provide an automatic volume controlling apparatus capable of automatically controlling the output volume of the loudspeaker by controlling the amplification degree of the volume depending on the level of the sound reproduced by the loudspeaker.

It is a third object of the invention to provide an automatic volume controlling apparatus capable of maintaining an appropriate volume over the entire frequency band of the sound signal by appropriately compensating the decrease of auditory volume due to the ambient noise when reproducing music.

The above and further objects and features of the invention will more fully be apparent from the following detailed description with accompanying drawings.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a block diagram of a conventional automatic volume controlling apparatus of the (c) type;

FIG. 2 is a graph showing an example of a result of the volume control by the conventional apparatus of the (c) type;

FIG. 3 is a block diagram showing another conventional automatic volume controlling apparatus;

FIG. 4 is a diagram explanatory of an operation of the conventional automatic volume controlling apparatus of FIG. 3;

FIG. 5 is a block circuit diagram of an automatic volume controlling apparatus in Embodiment 1 of the invention;

FIG. 6 is a graph showing an example of a result of the volume control in Embodiment 1 of the invention;

FIG. 7 is a block circuit diagram provided with delay means in Embodiment 1 of the invention;

FIG. 8 is a block circuit diagram of an automatic volume controlling apparatus in Embodiment 2 of the invention;

FIG. 9 is a block circuit diagram of an automatic volume controlling apparatus in Embodiment 3 of the invention;

FIG. 10 is a flowchart showing an operation of a DSP for estimating the level of the ambient noise in Embodiment 3 of the invention;

FIG. 11 is a flowchart showing an operation of a microcomputer in Embodiment 3 of the invention;

FIG. 12 is a flowchart showing an operation microcomputer in Embodiment 3 of the invention;

FIG. 13 is a block diagram showing an automatic volume controlling apparatus in embodiment 4 of the invention;

FIG. 14 is a diagram explanatory of an operation of the automatic volume controlling apparatus in Embodiment 4 of the invention;

FIG. 15 is a block diagram showing an automatic volume controlling apparatus in Embodiment 5 of the invention;

FIG. 16 is a block diagram provided with delay means before sound signal analyzing means in Embodiment 5 of the invention;

FIG. 17 is a block diagram provided with delay means before sound signal filtering means in Embodiment 5 of the invention;

FIG. 18 is a block diagram showing an automatic volume controlling apparatus in Embodiment 6 of the invention;

FIG. 19 is a block diagram showing the configuration of adaptive filtering means in Embodiment 6 of the invention; and

FIG. 20 is a block diagram having delay means in Embodiment 6 of the invention.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

[Embodiment 1]

FIG. 5 is a block circuit diagram showing Embodiment 1 of the invention, in which numeral **101** denotes a sound source, **102, 103** denote control amplifiers, **104, 105, 106** denote control amplifiers, **107** and **108** denote loudspeakers, **109** denotes a microphone, **110, 111, 112** denote A/D converters, **113, 114** denote adaptive filters, **115** denotes a subtracter, **116, 117** denote level judging devices for level converting A/D converted signals, and comparing the level with the reference level, **118** denotes a signal converter for level converting the output of the subtracter **115**, and **119** denotes an amplification controller for sending an amplification control signal corresponding to the output of the signal converter **118** to the control amplifiers **102, 103**.

In the automatic volume controlling apparatus according to the present invention, the right and left channel signals from the sound source **101** pass through the control amplifiers **102, 103**, and are then fed to the amplifiers **104, 105** and to the A/D converters **110, 111**. The outputs from the amplifiers **104, 105** are inputted to the loudspeakers **107, 108**, respectively, and emitted as sound waves. The ambient noise and speaker produced sound picked up by the microphone **109** are fed into the A/D converter **112** through the amplifier **106**.

The output digital signals from the A/D converters **110, 111** are fed into the adaptive filters **113, 114**, respectively, to be filtered, and their outputs are fed into the subtracter **115**. The output digital signal from the A/D converter **112** is also fed to the subtracter **115**, and the result of subtracting both the outputs of the adaptive filters **113, 114** from the output of the A/D converter **112** is fed to the signal converter **118**, and is also fed to the adaptive filters **113, 114** as coefficient updating signals.

The filtering coefficient of the adaptive filter **113** is updated so as to bring the signal from the A/D converter **110** closer to the signal which passes through the control amplifier **102** to the amplifier **104**, is emitted from the loudspeaker **107** as a sound wave, is picked up by the microphone **109**, and which reaches the A/D converter **112** via the amplifier **106**. The filtering coefficient of the adaptive filter **114** is updated so as to bring the signal from the A/D converter **111** closer to the signal which passes through the control amplifier **103** to the amplifier **105**, is emitted from the loudspeaker **108** as a sound wave, is picked up by the microphone **109**, and which reaches the A/D converter **112** via the amplifier **106**.

Herein, the coefficient updating algorithm for asymptotically minimizing an error signal from the subtracter **115** is known. As a result, the speaker sound component detected by the microphone **109** is canceled, and a correct ambient noise is obtained.

The input to the signal converter **118** is converted into a direct current with a time constant suitable for man's hearing characteristics (for example, a time constant used in a sound level meter), and is further logarithmically converted and outputted to the amplification controller **119** as a decibel value. Accordingly, an amplification control signal is sent from the amplification controller **119** to the control amplifiers **102, 103**.

The outputs of the A/D converters **110, 111** are sent, to the adaptive filters **113, 114**, and also to the level judging devices **116, 117** to be converted to direct currents with a suitable time constant (e.g., approximating the time constant of the sound level meter) and compared with the reference level. When the output signal from the A/D converters **110, 111** are smaller than the reference level, signals are independently sent to the adaptive filters **113, 114** to control the adaptive filters **113, 114** to maintain the coefficients without updating. As a result, if one of the input levels of the adaptive filters **113, 114** is smaller than the reference level, only the one filter with a larger input level is actuated, while the other does not affect the estimated value. When both the inputs are smaller than the reference level, the coefficients are not updated to extraordinary values, then when a signal of a normal level is fed next time, filtering and updating of the coefficients are executed under the optimum conditions.

In this apparatus, when a signal is not outputted from the sound source **101**, the level of the ambient noise is obtained as the output of the signal converter **118**. When a signal is outputted from the sound source **101**, an estimated value of the ambient noise level in the embodiment is obtained as the output of the signal converter **118**. At this time, when the signal picked up by the microphone **109** is level converted, the level of a mixed sound of the speaker produced sound with the ambient noise is obtained.

FIG. 6 is a graph showing an example of a result obtained by the apparatus shown in FIG. 5, in which A indicates the level of the mixed sound of the ambient noise with speaker produced sound, which is picked up by the microphone, D indicates the estimated level of the ambient noise detected in the embodiment, and C indicates the actual level of the ambient noise picked up by the microphone without producing the speaker sound. The ambient noise herein is imitatively reproduced and added by the loudspeaker.

It is apparent from FIG. 6 that the level of the ambient noise is correctly obtained in the embodiment if the signal of the sound apparatus is stereo.

Incidentally, delay means may be also provided for suspending the control for an appropriate time after the start of the operation of the adaptive filters **113, 114** since converting the adaptive filters **113, 114** to some degree requires some time. FIG. 7 is a block diagram showing the configuration of providing delay means **130, 131** after the adaptive filters **113, 114**, respectively, and the same parts as in FIG. 5 are identified with the same reference numerals and their explanations are omitted. When starting to operate, the adaptive filters **113, 114** send the signal noticing the start of the operation to the delay means **130, 131**. On receiving the signal, the delay means **130, 131** suspend the control by the control amplifiers **102, 103** for an appropriate time, and

then send a control start signal to the amplification controller **119**, so that the volume is not controlled for a certain time until converging to a certain extent after the start of the operation of the adaptive filters **113, 114**, thereby preventing faulty operation.

In Embodiment 1, the adaptive filter **113** through the amplification controller **119** are described as independent devices, but the process by those devices may be realized by a program process using a digital signal processor (DSP).

[Embodiment 2]

FIG. **8** is a block circuit diagram showing an embodiment of the automatic volume controlling apparatus of the invention, in which numeral **201** denotes a sound source, **202, 203** denote control amplifiers, **204, 205, 206** denote amplifiers, **207, 208** denote loudspeakers, **209** denotes a microphone, **210** denotes a resistance attenuator, **211, 212, 213** denote A/D converters, **214, 215** denote adaptive filters, **216** denotes a subtracter, **217** denotes an adder, **218, 219, 220** denote signal converters for level converting A/D converted signal, **221** denotes an amplification controller for sending an amplification control signal depending on the outputs of the signal converters **219, 220** to the control amplifiers **202, 203**, **222** denotes a reference signal generator, **223** denotes an attenuation controller for operating the reference signal generator **222** and sending an attenuation control signal corresponding to the output of the signal converter **218** at this time to the resistance attenuator **210**, and **224** denotes a coefficient holder for holding the determined attenuation coefficient.

In this automatic volume controlling apparatus, the right and left channels' signals from the sound source **201** are fed both to the amplifiers **204, 205**, and to the A/D converters **212, 213** through the control amplifiers **202, 203**. The outputs from the amplifiers **204, 205** are supplied to the loudspeakers **207, 208**, respectively, and are emitted as sound waves. The ambient noise plus speaker produced sound picked up by the microphone **209** are fed into the A/D converter **211** through the amplifier **206** and the resistance attenuator **210**.

On the other hand, the output digital signals from the A/D converters **212, 213** are fed into the adaptive filters **214, 215** to be filtered, and the filtered outputs are fed to the subtracter **216** and adder **217**. The output digital signal from the A/D converter **211** is also fed into the subtracter **216**, from which the outputs of the both adaptive filters **214, 215** are subtracted. The subtracted result is fed into the signal converter **219**, and is also fed to the adaptive filters **214, 215** as the coefficient updating signals.

The outputs of the adaptive filters **214, 215** are also fed to the adder **217**, and their sum is fed into the signal converter **220**.

Accordingly, the coefficient of the adaptive filter **214** is updated so as to bring the signal from the A/D converter **212** closer to the signal which passes through the control amplifier **202** to the amplifier **204**, is emitted from the loudspeaker **207** as a sound wave, is picked up by the microphone **209**, and which reaches the A/D converter **211** through the amplifier **206** and the resistance attenuator **210**. The coefficient of the adaptive filter **215** is updated so as to bring the signal from the A/D converter **213** closer to the signal which passes through the control amplifier **203** to the amplifier **205**, is emitted from the loudspeaker **208** as a sound wave, is picked up by the microphone **209**, and which reaches the A/D converter **211** through the amplifier **206** and the resistance attenuator **210**.

Herein, the coefficient updating algorithm for asymptotically minimizing an error signal from the subtracter **216** is known. As a result, the speaker sound component detected by the microphone **209** is canceled, and a correct ambient noise is obtained.

The inputs to the signal converters **219, 220** are respectively converted into direct currents with a suitable time constant (e.g., approximating the time constant of the sound level meter) and are further logarithmically converted and outputted to the amplification controller **221** as decibel values. Accordingly, an amplification control signal is sent from the amplification controller **221** to the control amplifiers **202, 203**.

On the other hand, the output of the A/D converter **211** is sent both to the subtracter **216** and the signal converter **218**. The signal converter **218** converts the output of A/D converter **211** into a direct current with a time constant approximating the time constant of the sound level meter, and further logarithmically converts and sends this direct current to the attenuation controller **223** as a decibel value.

The attenuation controller **223** usually sets the attenuation coefficient of the resistance attenuator **210** at that held by the coefficient holder **224**. When gain control is requested according to the input signal to the microphone **209**, the controller **223** causes the reference signal generator **222** to generate an adjustment signal. At this time, the attenuation amount of the resistance attenuator **210** is controlled so that the level of the signal emitted as sound wave by the loudspeakers **207, 208** from the control amplifiers **202, 203** through the amplifies **204, 205**, picked up by the microphone **209** and inputted to the attenuation controller **223** through the resistance attenuator **210**, A/D converter **211** and signal converter **218** may settle within an appropriate range, while the attenuation coefficient is held in the coefficient holder **224**.

[Embodiment 3]

In Embodiment 2, the adaptive filter **214** through the amplification controller **221** are described as independent devices, but the process by the devices may be realized by a program process using a DSP.

FIG. **9** is a block circuit diagram showing another embodiment of the automatic volume controlling apparatus of the invention, in which the same parts as in FIG. **8** are identified with the same reference numerals and their explanations are omitted herein. In the figure, numeral **225** denotes a DSP for volume amplification, **226** denotes a DSP for detecting the ambient noise level and sound reproduction level by the same processing in the devices **214** through **220** in FIG. **8**, **227** denotes a microcomputer, and **228** denotes a nonvolatile memory.

The operation in Embodiment 3 will be explained below with reference to FIGS. **10** and **11**. FIG. **10** is a flowchart showing an operation by a program stored in the DSP **226**, and FIGS. **11** and **12** are flowcharts showing an operation by a program stored in the microcomputer **227**.

In this automatic volume controlling apparatus, the right and left channels' signals from the sound source **201** are fed into the amplifiers **204, 205** and the A/D converters **212, 213** via the DSP **225** for volume amplification. The outputs from the amplifiers **204, 205** are supplied to the loudspeakers **207, 208**, respectively, and are emitted as sound waves. The ambient noise plus

speaker produced sound picked up by the microphone 209 are fed into the A/D converter 211 via the amplifier 206 and the resistance attenuator 210.

The DSP 226 for estimating the noise level first receives the data from the A/D converters 211 through 213 when the power source is turned on as shown in FIG. 10 (S1). Then, the data of the sound reproduced by the loudspeakers 207, 208, which is received from the A/D converter 212 and A/D converter 213 are filtered by variable coefficient FIR digital filters (S2). The estimated value of the noise level is obtained by subtracting the data of the filtering result from the data of the sound picked up by the microphone 209, which is received from the A/D converter 211 (S3). By making use of the subtracted result, the coefficient of the variable coefficient filter is updated so as to bring the signals from the A/D converters closer to the signals which pass through the amplifiers 204, 205, are emitted from the loudspeakers 207, 208 as sound waves, picked up by the microphone 209, and which reach the A/D converter 211 through the amplifier 206 and the resistance attenuator 210 (S4). Further, the filtering results are summed up to detect the speaker sound level (S5). The subtracted result, the added result, and the input data from the A/D converter 211 are level converted and sent to the microcomputer 227 as a noise level, a sound reproduction level, and a microphone signal level, respectively.

Herein, the coefficient updating algorithm for asymptotically minimizing an error signal in the subtraction is known. As a result, the speaker sound component detected by the microphone 209 is canceled, and a correct ambient noise is obtained.

When the power source is turned on as shown in FIGS. 11 and 12, the microcomputer 227 first sets the attenuation amount of the resistance attenuator 210 to the value being read out from the nonvolatile memory 228 (S11). On receiving the three data, that is, the noise level, sound reproduction level, and microphone output signal level from the DSP 226 (S12), the amplification control amount is calculated from the noise level, the sound reproduction level and the set value of attenuation of the resistance attenuator 210 (S13). The amplification control amount is transmitted to the DSP 225 to vary the volume (S14). Whether or not to set the attenuation amount of the resistance attenuator 210 newly is checked (S15), and if not, the next set of data is received from the DSP 226, and the same steps S13, S14 are repeated.

When newly setting the attenuation of the resistance attenuator 210, the microcomputer 227 sends an instruction to the DSP 225 to out, put the adjustment signal to the resistance attenuator 210 (S16). When receiving the data from the DSP 226 (S17), the attenuation amount of the resistance attenuator 210 is changed depending on the microphone output signal level, which is repeated until the microphone signal level settles within an appropriate range (S17–S19). When the microphone signal level settles within an appropriate range, the microcomputer 227 instructs the DSP 225 to stop to output the adjustment signal (S20), and stores a newly set value of the attenuation amount of the resistance attenuator 210 in the nonvolatile memory 228 (S21), so that, from the next time, this value is set when the power source is turned on.

The mathematical processing for generating the adjustment signal (for example, random noise) is known, and it is not particularly difficult to execute the process by the DSP.

[Embodiment 4]

FIG. 13 is a block diagram showing a fourth embodiment of the automatic volume controlling apparatus of the invention. In the drawing, numeral 301 denotes an input terminal, 302 denotes an output terminal, 303 denotes an A/D converter, 304 denotes an FIR filter, 305 denotes a D/A converter, 306 denotes a first fast Fourier transformer 306 (FFT), 307 denotes a level detector, 308 denotes gain calculating means, 309 denotes an interpolator, 310 denotes filtering coefficient calculating means, 311 denotes a microphone, 312 denotes an amplifier, 313 denotes an A/D converter, 314 denotes a second FFT, 315 denotes a level detector, and 318 denotes an A/D converter. Moreover, 340 denotes sound signal analyzing means, 341 denotes ambient noise analyzing means, and 342 denotes filtering means. In this embodiment, it is assumed that the output terminal 302 is like an earphone so that the reproduced sound is hardly picked up by the microphone 311.

The inputted sound signal from the input terminal 301 is converted into a digital signal by the A/D converter 318, and is transformed by the first FFT 306 into a set of data for supplying levels of signal components of the center frequencies of the bands at equal frequency intervals. Thus the signal component level in an analysis period of every frequency band is obtained. The signal component level is equally divided into n equal bands of half the sampling frequency according to the inputted number n for the sampling data. The fast Fourier transforming method is known, and may be executed by a DSP in practical use.

Since masking of a sound signal component (spectrum) in a certain frequency is determined by the noise level included in the critical band width centered on the certain frequency, it is desired to determine the analysis frequency band of sound signal and noise by finely dividing their frequency bands to the extent of the critical band width. But the critical band width is narrow in low frequency while broad in high frequency, so that the frequency interval of the data obtained by the fast Fourier transform with equal frequency intervals is too dense in high frequency range, if the frequency interval is proper in low frequency.

The level detector 307 provides the level of the sound signal contained in each frequency band by dividing the output of the FFT 306 in appropriate frequency bands, and processes to obtain an effective value by the entire data included in the established frequency band. A satisfactory auditory performance is achieved by dividing the sound frequency band (20 Hz to 200000 Hz) into about ten bands for every octave. Incidentally, it is known that the general noise spectrum is high in level in low frequency, while low in level in high frequency (the common household noise decreases at a rate of −6 dB/oct, or the automobile internal noise at a rate from −10 dB to −12 dB per octave). Accordingly, the processing amount in the FFT may be decreased by practically narrowing the detection range in the higher frequency of the sound signal level and ambient noise level.

In the embodiment, the FFT 306 and the level detector 307 compose the sound signal analyzing means 340, but they may be replaced by plural band filters with the number thereof corresponding to that of the required

frequency bands, and a level detector for detecting and smoothing the output.

The ambient noise signal obtained from the microphone **311** is amplified by the amplifier **312**, converted into a digital signal by the A/D converter **313**, and is transformed into level data at equal frequency intervals by the FFT **314**. The level detector **315** divides the output of the FFT **314** into the appropriate number of the frequency bands to provide the level of the ambient noise signal contained in every frequency band.

In the embodiment, the FFT **314** and the level detector **315** compose the ambient noise analyzing means **341**, but they may be replaced by plural band filters with the number thereof corresponding to that of the required frequency bands, and a level detector for detecting and smoothing the output.

On receiving the data output from the sound signal analyzing means **340** and the data output from the ambient noise analyzing means **341**, the gain calculating means **308** determines the gain of each frequency band.

The concept of gain calculation will be explained below. The decrease of auditory volume of the sound signal due to the masking by the ambient noise is considered as the phenomenon that the origin of the sensation level in the unit of sone (hereinafter referred to as the auditory volume) shifts to the masking level (unit: sone). That is, assuming Sm to be the auditory volume with masking, S to be the auditory volume without masking, and Sth to be the auditory volume equivalent to a masking level, the following equation is satisfied.

$$Sm=S-Sth=K(I^{\alpha}-Ith^{\alpha}) \tag{1}$$

wherein I expresses sound intensity (unit: W/m^2), Ith expresses sound intensity K equivalent to the masking level, and α is a constant depending on a frequency.

The relationship between the sound pressure level P (unit: dBspl) and the sound intensity I is expressed as follows in case of a plane wave.

$$P=10\cdot\log(I)+120 \tag{2}$$

FIG. **14** is a diagram showing the relation between the noise level and sound level, in which the curve **345** indicates the auditory volume with the ambient noise, the curve **346** indicates the masking level due to the ambient noise, and the curve **347** indicates the auditory volume without the ambient noise. Hence, in order to obtain the same auditory volume in case of having the ambient noise as in the absence of the ambient noise, it is necessary to shift the sound pressure level of the sound signal from the curve **347** to the curve **345** as indicated by an arrow in the diagram.

The gain calculating means **308** calculates the gain in each frequency band analyzed by the sound signal analyzing means to be given in order to equalize the auditory volume of the sound signal to that without the ambient noise, on the basis of the relation in FIG. **14**. More specifically, first a level of a reproduced sound P_0 in the case without volume control is estimated from the output of the sound signal analyzing means **340**, then a masking level Pt is estimated from the ambient noise level obtained from the output of the ambient noise analyzing means **341**, and finally a level of a reproduced sound P_1 to compensate for the decrease of the auditory volume due to the masking is calculated by the following equation which is derived from the above explanatory formulas.

$$P_1=P_0+10\cdot\log\{1+10^{(Pt-P_0)\alpha/10}\}/\alpha \tag{3}$$

Herein, the gain to be given to the sound signal is equivalent to P_1-P_0. The value of α is approximately 0.3 per a pure sound of 1 kHz. Though α takes different values among frequencies, the value 0.3 is practically suitable for the calculation except in case of a low frequency band. For example, when dividing the frequency band of the sound into ten bands by every octave, the reproduced sound levels are calculated by substituting individual values in α as for three low frequency bands including 100 Hz or less, while substituting 0.3 in α as for the other seven bands. The calculation of the formula (3) may also be achieved by approximate calculation or by a table search according to a conversion table of the relation among P_0, Pt, and P_1, or interpolation processing by a DSP or a microcomputer.

As clear from the above explanation, the gain calculating means **308** outputs the gain to be given to each frequency band of the sound signal, and the subsequent interpolator **309**, filtering coefficient calculating means **310**, and FIR filter **304** compose the filtering means **342** for giving the gain-to-frequency characteristic to the sound signal.

It is known that the coefficient of the FIR filter is nearly the desired filtering impulse response. It is also known that the frequency response and the impulse response in an arbitrary filtering can be mutually converted by Fourier transform. It is also known to obtain a characteristic approximate to a desired frequency response by inverse Fourier transform of an arbitrary frequency response to obtain the impulse response, and use it as the FIR coefficient.

Herein, the filtering coefficient calculating means **310** processes the output of the interpolator **309** by inverse Fourier transform to determine the FIR filtering coefficient, and sends the coefficient to the FIR filter **304**. At this time, the input to be given to the filtering coefficient calculating means **310** is the gain for each frequency being divided into n equal bands of half the sampling frequency of the sound signal, assuming the number of FIR filtering stages to be n. The interpolator **309** converts the gain determined in every, nearly the same, specific band width to the auditory characteristics in the gain calculating means **308** under the above-mentioned condition, wherein the gain for each frequency is divided to n equal bands of half the sampling frequency of the sound signal. The practical numerical processing method is known, which can be realized by a DSP or a microcomputer.

[Embodiment 5]

FIG. **15** is a block diagram showing a fifth embodiment of the an automatic volume controlling apparatus of the invention. In the diagram, numeral **301** denotes an input terminal, **302** denotes an output terminal, **303** denotes an A/D converter **304** denotes an FIR filter, **305** denotes a D/A converter, **306** denotes an FFT, **307** denotes a level detector, **308** denotes gain calculating means, **309** denotes an interpolator, **310** denotes filtering coefficient calculating means, **311** denotes a microphone, **312** denotes an amplifier, **313** denotes an A/D converter, **314** denotes an FFT, **315** denotes a level detector, **316** denotes a comparator, and **318** denotes an A/D converter. Further, numeral **340** denotes sound signal analyzing means, **341** denotes ambient noise analyzing means, and **342** denotes filtering means.

In this embodiment, as compared with Embodiment 4 where the reproducing means of the sound signal is an earphone or the like by which the reproduced sound is hardly picked up by the microphone for picking up the ambient noise, the sound reproducing means is a loudspeaker or the like, and the reproduced sound is picked up by the microphone together with the ambient noise.

As in Embodiment 4, the FFT 306 outputs the sound signal component levels of frequency bands at equal frequency intervals, and the FFT 314 outputs the ambient noise signal component levels of the frequency bands at the same equal frequency intervals as in the FFT 306. In this embodiment, the comparator 316 compares the identical frequency component levels of the FFTs 306 and 314, and the corresponding output of the FFT 314 is directly outputted when the output level of the FFT 314 is sufficiently higher than the output level of the FFT 306, which is the estimated level of the sound signal component reproduced by the loudspeaker or the like and picked up by the microphone. If not, one of the following processings is performed: output the previously outputted value again (previous value hold); output a prescribed value for each band; output the same value as the smaller one of the levels of the adjacent bands; output the value one prescribed value smaller than the previous outputted value. By this processing, a subsequent erroneous volume control on a mistake of the reproduced sound from the loudspeaker or the like for the noise can be avoided.

The reason that the ambient noise is detected by such processing of the comparator 316 will be explained below. In the FFTs 306 and 314, the frequency band is subdivided, so that a high sound signal level scarcely appears in a specific band. On the other hand, the ambient noise often has a continuous frequency spectrum, so that a certain level often appears in each of the divided bands. Accordingly, when the sound signal level is sufficiently low, the ambient noise is surely detected. In several bands, moreover, if the above-mentioned processing is done, even though the noise level cannot be detected, the correct noise level is probably obtained, on the whole.

In this embodiment, the outputs of the FFTs 306 and 314 are compared by the comparator 316 to detect the ambient noise level, but if the produced sound level is approximately known beforehand, (by comparing the output of the FFT 306 with the prescribed value in each band) the output of the FFT 314 may be directly outputted when the output of the FFT 306 is smaller than the prescribed value. If the output of the FFT 306 is larger than the prescribed value, one of the following processings is executed: output the previous output again (previous value hold); output a specified value in each band; output the same value as the smaller one of the levels of the adjacent bands; output the value one prescribed value smaller than the previous outputted value.

In Embodiment 5, in order to eliminate the effect of the reproduced sound signal into the microphone 311, the outputs of the FFTs 306 and 314 are compared by the comparator 316. But, since the sound signal into the microphone 311 propagates through the space from the electro-sound converting means such as loudspeaker before reaching the microphone 311, there occurs a time delay corresponding to the distance between the loudspeaker and microphone 311. To avoid the mistake of the sound signal for the ambient noise, it is necessary to compare the levels of the sound signals contained in the outputs of the FFTs 306 and 314 in the same timing by delaying the outputs from the FFT 306. FIG. 16 is a block diagram showing the configuration with delay means for providing the first FFT 306 an appropriate delay time, and the same parts as in FIG. 15 are identified with the same reference numerals and explanations are omitted. In case where the distance between the microphone and loudspeaker is far, the levels of the sound signals contained in the outputs of the FFTs 306 and 314 may be compared in the same timing with the FFT 306 by applying the delay corresponding to this distance to the route on the sound signal level detection side, that is, to the FFT. 306.

In detecting the levels of the sound signal and ambient noise by the fast Fourier transform, since it is necessary to analyze the data block in a definite time relating to the frequency resolution, it takes a certain time to obtain the result to delay the volume control by this certain time. The volume is controlled by the sound signal level and ambient noise level, as mentioned above, and usually delay in following up the ambient noise level does not matter too much. However, if the gain decrease control delays when the sound signal level suddenly raises, the sound is temporarily reproduced at an unnecessarily high level, which causes great discomfort to listeners. To avoid this, it is proper to provide delay means 332 for delaying the volume control by the time corresponding to this detection delay time before the filtering means 342 on the sound signal route (see FIG. 17). Yet, since the delay given by the delay means 332 is equal to the time where a delay occurring by the processing in the filtering means 342 is subtracted from the required delay, an extra delay is unnecessary if the delay by the FIR filter 4 is sufficiently large.

[Embodiment 6]

FIG. 18 is a block diagram showing a sixth embodiment of the automatic volume controlling apparatus of the invention. In the diagram, numeral 301 denotes an input terminal, 302 denotes an output terminal, 303 denotes an A/D converter, 304 denotes an FIR filter, 305 denotes a D/A converter, 306 denotes an FFT, 307 denotes a level detector, 308 denotes gain calculating means, 309 denotes an interpolator, 310 denotes filtering coefficient calculating means, 311 denotes a microphone, 312 denotes an amplifier, 313 denotes an A/D converter, 314 denotes an FFT, 315 denotes a level detector, 316 denotes a comparator, 317 denotes adaptive filtering means, and 318 and 319 denote A/D converters. Numeral 340 denotes sound signal analyzing means, 341 denotes ambient noise analyzing means, and 342 denotes filtering means.

Herein, the parts having the same reference numerals as in Embodiments 4 and 5 operate in the same manner. In this embodiment, same as in Embodiment 5, the sound reproducing means is a loudspeaker or the like, and the reproduced sound is picked up by the microphone together with the ambient noise. The adaptive filter 317, same as in Embodiment 5, receives both the ambient noise signal obtained by amplifying and A/D converting the output of the microphone 311, and the sound signal having a correlation as high as possible with the sound signal reproduced by the loudspeaker (hereinafter referred to as a reference sound signal; in FIG. 18, the output of the D/A converter 305 converted into a digital signal by the A/D converter 319 is used as the reference sound signal). The adaptive filter

317 removes the component correlated with the reference sound signal, that is, the reproduced sound component by the loudspeaker from the mixed signal of the sound and ambient noise or the like, so as to output only the ambient noise.

FIG. 19 is a block diagram showing the constitution of the adaptive filter 317 in the sixth embodiment of the invention, in which numeral 323 denotes a variable coefficient FIR filter, 324 denotes a subtracter, and 325 denotes coefficient updating means. The adaptive filter 317 in this constitution is known. The subtracter 324 subtracts the reference sound signal passing through the variable coefficient FIR filter 323 from the mixed signal of the sound and ambient noise. The coefficient updating means 325 updates the coefficient of the variable coefficient FIR filter 323 so as to minimize an error signal outputted from the subtracter 324. By this coefficient updating, the variable coefficient FIR filter 323 consequently imitates the transmission characteristics of the route through the D/A converter to the power amplifier via the output terminal 302, the electro-acoustic converter such as the loudspeaker, sound emitted space, and the microphone (where the sound signal obtained from the microphone passes after the reference sound signal pickup point). In other words, the variable coefficient FIR filter 323 transforms the reference sound signal to be equivalent to the sound signal component contained in the mixed signal of the sound and ambient noise, by convolving the characteristic which imitates the transmission characteristic of the route from reproduction to pickup of the sound signal into the reference sound signal.

As mentioned above, when detecting the levels of the sound signal and ambient noise by the fast Fourier transform, it takes a certain time to obtain the result. This delay occurs due to analyzing the data block of a definite time relating to the frequency resolution. Because of this delay, the sound is temporarily reproduced at a level higher than necessary, which causes great discomfort to listeners. To avoid this, it is proper to delay the volume control by the time corresponding to this detection delay before the filtering means 342 on the sound signal route. FIG. 20 is a block diagram of the constitution with delay means 331 before the filtering means 342, and the same parts as in FIG. 18 are identified with the same reference numerals and explanations are omitted. The delay means 331 gives a signal to the filtering means 342 by delaying the timing of the volume control to prevent the sound from being temporarily reproduced at a higher level than necessary. However, since the delay given by the delay means 331 is equal to the time where delay occurring in the filtering means 342 is subtracted from the required delay, an extra delay is unnecessary if the delay in the FIR filter 4 is large.

In the embodiments 4 through 6, the FIR filter 304, FFT 306, level detector 307, gain calculating means 308, interpolator 309, filtering coefficient calculating means 310, level detector 315, and adaptive filter 317 are mentioned as independent devices, but the processes by those devices may be sequentially executed and processed by the DSP. Besides, if all of the processes cannot be executed by a single DSP due to the restriction in capabilities or the like, it is possible to distribute and process by plural DSPs or microcomputers.

In such cases, the A/D converters 313, 318, 319 may time share a single element.

As this invention may be embodied in several forms without departing from the spirit of essential characteristics thereof, the present embodiments are therefore illustrative and not restrictive, since the scope of the invention is defined by the appended claims rather than by the description preceding them, and all changes that fall within metes and bounds of the claims, or equivalence of such metes and bounds thereof are therefore intended to be embraced by the claims.

What is claimed is:

1. An apparatus for automatically controlling a volume of sound produced by loudspeakers, depending on the volume of ambient noise, comprising:
  - a microphone for picking up 2-channel mixed sounds including sound produced by the loudspeakers and ambient noise;
  - adaptive filtering means having variable filtering coefficients, receiving a same set of 2-channel input signals that are inputted to the loudspeakers, for adaptively filtering the 2-channel input signals;
  - subtracting means for respectively subtracting outputs of the adaptive filtering means, corresponding to the 2-channel sounds, from the mixed sounds including the sounds from the loudspeakers and the ambient noise;
  - means, responsive to outputs from the subtracting means, for updating the filtering coefficients of the adaptive filtering means to minimize the output signal from the subtracting means; and
  - means for controlling the volumes of the loudspeakers depending on the outputs from the subtracting means.

2. An automatic volume controlling apparatus of claim 1, further comprising means for stopping an update of the filtering coefficients of the adaptive filtering means when a level of the 2-channel mixed sounds is lower than a reference level.

3. An automatic volume controlling apparatus of claim 1, further comprising means for delaying a start of volume control after the adaptive filtering means starts to operate until the adaptive filtering means converges.

4. An automatic volume controlling apparatus of claim 1 wherein the means for controlling the volumes includes a digital signal processor.

5. An automatic volume controlling apparatus of claim 1 wherein the adaptive filtering means includes a digital signal processor.

6. An automatic volume controlling apparatus of claim 1, further comprising:
  - means for attenuating signals picked up by the microphone;
  - means for controlling a gain of the attenuating means according to the volume of the loudspeakers; and
  - means for holding the gain of the attenuating means.

7. An automatic volume controlling apparatus of claim 6 wherein the gain holding means holds the gain even after the power is cut off.

8. An automatic volume controlling apparatus of claim 6 wherein the gain controlling means includes a microcomputer.

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⑭発明の名称　受話音量調整方式

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明　細　書

1．発明の名称

受話音量調整方式

2．特許請求の範囲

(1)　受話器を含む受話回路と、スピーカを含むスピーカ回路とを備え、選択制御信号により前記受話回路又は前記スピーカ回路のいずれかを動作状態にして受話器受話又はスピーカ受話の状態を選択するとともに、前記スピーカ受話が選択されたときに前記スピーカの音量を調整するための音量調整手段を備えた受話音量調整方式において、前記音量調整手段となる音量調整用可変抵抗器を含む第1のルートと、第1の固定抵抗よりなる第2のルートと、前記第1のルートに含まれる抵抗の一部を除去するか別の第2の固定抵抗に置換した第3のルートとのいずれかを選択する手段を備え、前記スピーカ受話のときには前記制御信号により前記第1のルートが選択され、前記受話器受話のときは前記制御信号及び受話音量固定モードと受話音量可変モードとの選択用の別の制御信号により前記第2のルート又は前記第3のルートを選択するように構成されたことを特徴とする受話音量調整方式。

(2)　受話バッファ増幅回路の出力側に、受話器を含む受話回路と、スピーカを含むスピーカ回路とが接続され、選択制御信号により前記受話回路又は前記スピーカ回路のいずれかを動作状態にして受話器受話又はスピーカ受話の状態を選択するとともに、前記スピーカ受話が選択されたときに前記スピーカの音量を調整するための音量調整手段を備えた受話音量調整方式において、前記受話バッファ増幅回路が入力回路と帰還回路を含む演算増幅器よりなり、該演算増幅器には前記音量調整手段となる音量調整用可変抵抗器を含む第1のルートと、第1の固定抵抗よりなる第2のルートと、前記第1のルートに含まれる抵抗の一部を除去するか別の第2の固

定抵抗に置換した第3のルートとのいずれかを選択して前記帰還回路又は前記入力印加回路として接続する手段を備え、前記スピーカ受話のときには前記制御信号により前記第1のルートが選択され、前記受話器受話のときは前記制御信号及び受話音量固定モードと受話音量可変モードとの選択用の別の制御信号により前記第2のルート又は前記第3のルートを選択するように構成されたことを特徴とする受話音量調整方式。

3．発明の詳細な説明

（発明の技術分野）

本発明は、受話器音量をスピーカ音量調整用可変抵抗器で変化可能にしたボタン電話機に関するものである。

（従来技術とその問題点）

従来のボタン電話機では、601回路網のような局電源動作形の回路網を使用していた為、スピーカ増幅器等の回路とのグランドレベルが異なり、スピーカ増幅器側に接続されている音量調整用可変抵抗器を受話器側に自動的に接続替えすることが出来ず、音量調整用可変抵抗器による受話器音量の調節が出来なかった。一方、電子化された回路網とスピーカ増幅器等の回路のグランドレベルが一致した回路網を用いたボタン電話機でも、スピーカの音量可変幅は30倍程度、受話器側の音量可変幅は½～2倍程度とその要求される可変幅が異なる為、一般的には第1図のような回路構成をとっている。ここで、1はマイクロホン、4は送話器、2，5，7は送話増幅器、3，6は送話路スイッチ、8はハイブリッド回路、9は通話線端子、10，16，22は受話増幅器、11，19は受話路スイッチ、18はインバータ、12は音量切換スイッチ、13，14，15，21は固定抵抗、20は音量調整用可変抵抗器、17は受話器、23はスピーカである。40はマイクロホン1をオンオフ制御する制御信号、41は送話器4と受話器17をオンオフ制御し同時にスピーカ23をオンオフ制御する制御信号である。

このような回路構成から理解されるように、従来回路では、音量調整用可変抵抗器20による受話器音量の調節は出来なかった。このため、受話器音量の調節は通常、2段階程度の音量切換スイッチ12を受話増幅器16側に実装し、実現していた為、通話が終了するごとにスイッチ12を戻さなければならず、利用者の使用上不便であり、又、コスト高となっていた。

（発明の目的）

本発明はスピーカ音量と受話器音量とでその可変幅を任意に設定可能とし、且つ、受話器受話の初期状態には必ず固定音量となりしかもダイヤル又はボタンの操作で可変音量モードに切替えたときには受話器音量をスピーカ受話と同一の音量調整器で調整可能にした受話音量調整方式を提供するものである。

（実施例）

以下図面により本発明を詳細に説明する。

第2図(a)は本発明の一実施例である。ハンドセット通話時の受話信号は、通話線端子9より入力され、ハイブリッド回路8を通り、受話バッファ増幅回路10で増幅され、受話器通話路スイッチ11を通り、受話増幅器16で再び増幅され、受話器17より出力される。スピーカ受話時の受話信号は通話線端子9より入力されハイブリッド回路8を通り受話バッファ増幅回路10で増幅され、スピーカ通話路スイッチ19を通り、スピーカ増幅器22で再び増幅され、スピーカ23より出力される。

以上のように受話信号は、受話器17より出力される場合でも、スピーカ23より出力される場合でも、受話バッファ増幅回路10を通るため、受話バッファ増幅回路10に音量調整用可変抵抗器20を挿入し各々の状態においてその利得を変化することで各々の出力形態により、音量を変化することが出来る。以下に受話バッファ増幅回路10の利得について示す。受話バッファ増幅器10は本実施例では演算増幅器32で構成されており、信号源抵抗は抵抗24であり、帰還抵抗はルート1，ルート2及びルート3よりなり、そのルート1はスピーカ通話路を制御するスイッチ25と音量調整用可変抵抗器20と抵抗28との合成抵抗よりなり、ルート2は

受話音量可変時にオフになるスイッチ30と送話器・受話器通話路を制御するスイッチ31と抵抗29との合成抵抗よりなり、ルート3は受話音量可変時にオンになるスイッチ26と抵抗27と音量調整用可変抵抗器20と抵抗28との合成抵抗から構成されている。

各スイッチの動作条件は次の通りである。スイッチ25は受話信号がスピーカ23より出力される場合に制御信号41の反転信号によりオンになるスイッチ、スイッチ26は受話信号が受話器17より可変音量で出力される場合に制御信号42でオンになるスイッチ、スイッチ30は受話信号が受話器17より可変音量で出力される場合に制御信号42の反転信号でオフになるスイッチ、スイッチ31は受話信号が可変音量，固定音量のいかんにかかわらず、受話器17より出力される場合に制御信号41によりオンになるスイッチである。以上のように、制御信号41，42により、ボタン電話機の状態に応じスピーカ23より出力する場合はルート1が、受話器17より固定音量で出力する場合はルート2が、受話器17より可変音量で出力する場合はルート3が選択されるべく制御されるよう構成されている。これらのスイッチの制御は、CPU等からの制御で行われる為、受話音量可変モードでの通話がフックダウン等により終了した場合には自動的に通常の受話音量固定モード等に切換え可能である。演算増幅器32の反転増幅器の利得は、（帰還抵抗）／（信号源抵抗）で求められる。故に、ここで受話バッファ増幅回路10の利得は各スイッチのオン時の順抵抗が0Ωとすると、スピーカ23よりの出力の場合は帰還ルート1が選択され、音量調整用可変抵抗器20が最小の場合の(抵抗28)／(抵抗24)から最大の場合の{(音量調整用可変抵抗器20)+(抵抗28)}／(抵抗24)まで変化可能となり、(音量調整用可変抵抗器20)＝29×(抵抗28)とすることで、30倍の可変幅が確保出来る。受話器17より固定音量で出力する場合には、帰還ルート2が選択され(抵抗29)／(抵抗24)の利得が得られる。受話器17より可変音量で出力する場合には、帰還ルート3が選択され、音量調整用可変抵抗器20が最小の場合の{(抵抗27)+(抵抗28)}／(抵抗24)から最大の場合の{(音量調整用可変抵抗器20)+(抵抗27)+(抵抗28)}／(抵抗24)まで変化可能となり、ここで{(抵抗27)+(抵抗28)}＝½×(抵抗29){(音量調整用可変抵抗器20)+(抵抗27)+(抵抗28)}＝2×(抵抗29)とすることで、受話器17よりの固定音量出力に対し½倍～2倍まで変化可能となる。ルート1より(抵抗28)＝1/29×(音量調整用可変抵抗器20)であるから、(抵抗28)：(音量調整用可変抵抗器20)：(抵抗27)：(抵抗29)＝1：29：9.7：19.3とすることにより、スピーカ23からの出力時の可変幅が30倍に又受話器17からの出力可変幅½倍～2倍を達成することが出来る。

第2図(a)の実施例は第2図(b)又は(c)の回路に変更可能である。この場合には、第2図(b)はスイッチ25の制御により可変抵抗器20に抵抗34を並列に接続したり切り離すことにより、受話バッファ増幅回路10の可変抵抗器20による増幅幅を変えている。また、第2図(c)はスイッチ35の制御により可変抵抗器20の両端を開放または短絡することにより受話バッファ増幅回路10の増幅度が可変抵抗器20に依存するかしないかを決めている。

第2図(a)(b)(c)は、ルート1，2，3を受話バッファ増幅回路10における演算増幅器32の帰還ルートとして用いた場合の実施例であるが、第3図(a)はルート1，2，3を演算増幅器の入力回路として用いた場合の実施例である。ここでは、第2図(a)(b)(c)の帰還ルート1，2，3に関連する回路素子20，25，26，27，28，29，30，31，33と入力側の抵抗24とを入れ替えた構成をとっている。また、可変抵抗器20の変化による効果が第2図(a)(b)(c)の場合と逆になるので、図示のように、第3図(a)では第2図(a)(b)(c)の場合と接続極性を異ならせている。主要な動作は第2図(a)(b)(c)の実施例の場合と同様であるので、詳細な説明は省略する。

図示していないが、第2図(b)(c)における入力回路10aと帰還回路10bとを、第2図(a)から第3図(a)への変更と同様の要領により、置換する変形をすることができる。

さらに、第3図(a)の実施例は第3図(b)の回路に変更可能である。

第3図(b)は制御信号42の制御でスイッチ31またはスイッチ30のいずれかをオンにさせ、ハイブリッド回路8の出力を可変抵抗器20を通すか否かを選択し、スイッチ31がオンになり可変抵抗器20を通るルートが選択されたときは、更に制御信号41により、スイッチ25, 26のいずれかがオンになり、抵抗34, 27のいずれかが可変抵抗器20に接続され、可変抵抗器を通り演算増幅器に入力される信号レベルの変化範囲幅を変えている。

第3図(c)および(d)は第3図(a)および(b)において演算増幅器32と抵抗34を省き、ハイブリッド回路8の出力から受話器7あるいはスピーカ23までの増幅度が第3図(a)または(b)と同じになるように受話増幅器16とスピーカ増幅器22の増幅度を変えたものであり動作は実質上同様である。これらの第3図(c)(d)の実施例では前置増幅器の機能を有する演算増幅器32を用いていないため、主増幅器となるスピーカ増幅器22に大きい利得が必要であり、ハイブリッド回路8の出力からスピーカ増幅器22の入力までのラインを高インピーダンスに保つ必要性から、周囲からの誘導雑音を拾い易い欠点はあるが、増幅器の数を少なくすることができるという利点が得られる。

従って、第3図(c)(d)を除く他の実施例は、前置増幅器の機能を有する演算増幅器32を必要とするけれども、スピーカ増幅器22に対する所要利得の軽減とハイインピーダンス入力ラインの短縮化のために、誘導雑音の減少と利得配分上の設計容易性等の利得を得ることができる。

以上の実施例では、演算増幅器32の(−)側端子に信号を入力し、(+)側端子を接地する接続をとっているが、各実施例を第4図に示すように、(+)側端子に信号を入力し、(−)側端子に帰還信号を印加するように変形することができる。ここで、第2図(a)(b)(c)の実施例は帰還回路10bを制御するように構成され、第3図(a)(b)の実施例は入力回路10aを制御するように構成されていたことになる。

(発明の効果)

以上説明したように、本発明によれば、一つの音量調整用可変抵抗器によりスピーカ音量及び受話器音量が変化できる為、スイッチのように段階的で無く、受話器音量を連続的に調整して適正音量にセットすることが出来る。さらに、オンフックで自動的に受話音量可変状態が解除される為、通話終了時にスイッチ等を元に戻す必要が無くなる等の利点がある。

4. 図面の簡単な説明

第1図は従来のボタン電話機の構成例を示すブロック図、第2図(a)(b)(c)及び第3図(a)(b)(c)(d)は本発明の実施例を示すブロック図、第4図は本発明に用いられる受話バッファ増幅回路の変形例を示す回路略図である。

1…マイクロホン、 2…マイクロホン増幅器、 3…マイクロホン通話路を制御するスイッチ、 4…送話器、 5…送話増幅器、 6…送話器通話路を制御するスイッチ、 7…送話バッファ増幅器、 8…ハイブリッド回路、 9…通話線端子、 10…受話バッファ増幅回路、 11…受話器通話路を制御するスイッチ、 12…受話音量切換スイッチ、 13, 14, 15, 21…抵抗、 16…受話増幅器、 17…受話器、 18…インバータ、 19…スピーカ通話路を制御するスイッチ、 20…音量調整用可変抵抗器、 22…スピーカ増幅器、 23…スピーカ、 24, 27, 28, 29, 34…抵抗、 25, 26, 30, 31, 35…スイッチ、 32…演算増幅器、 33…インバータ。

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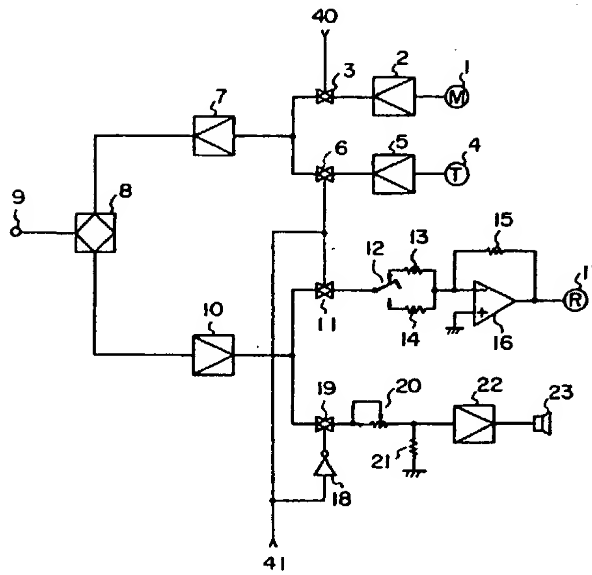
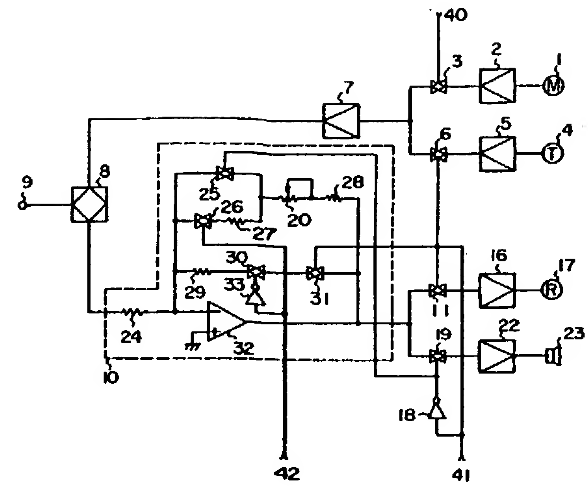
外1名

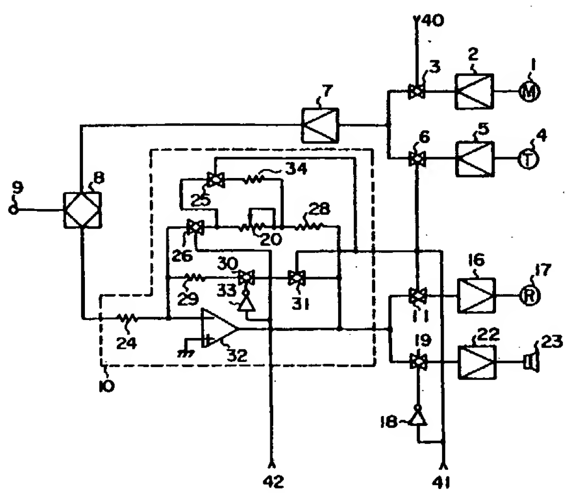
第1図



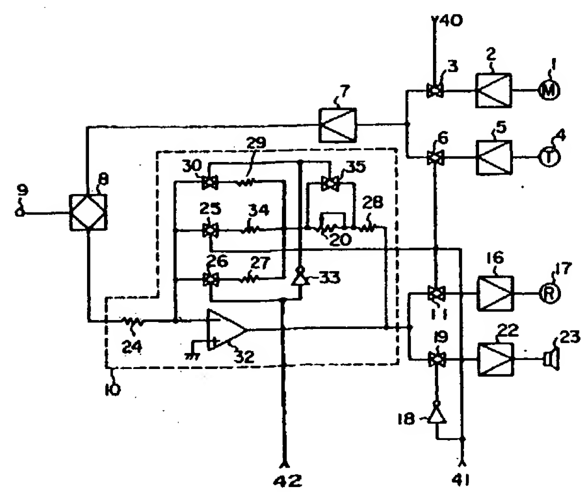
第2図 (a)



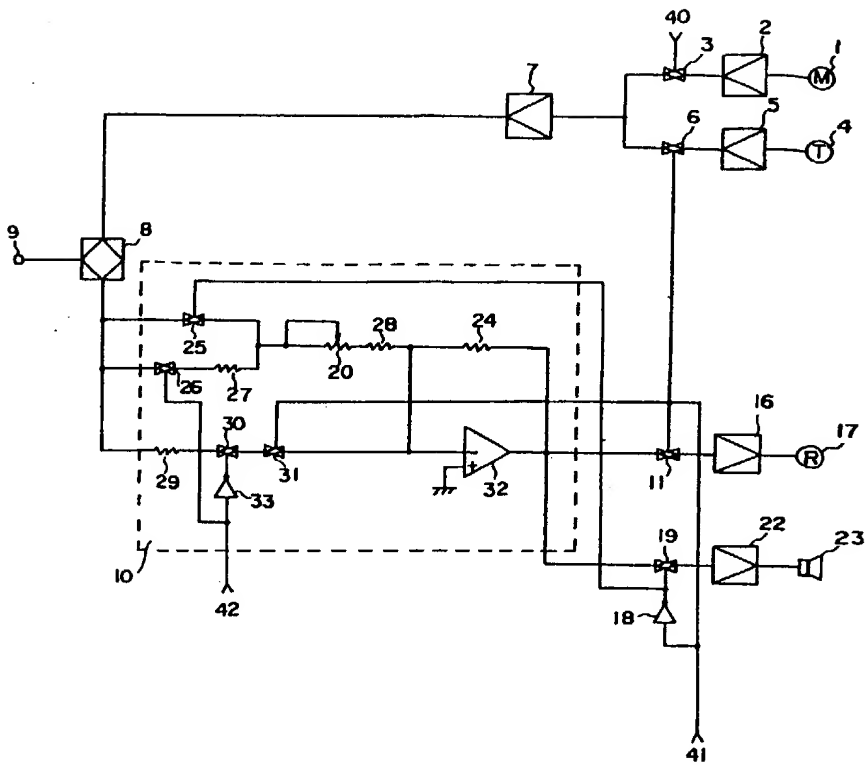
第2図 (b)



第2図 (c)



第3図(a)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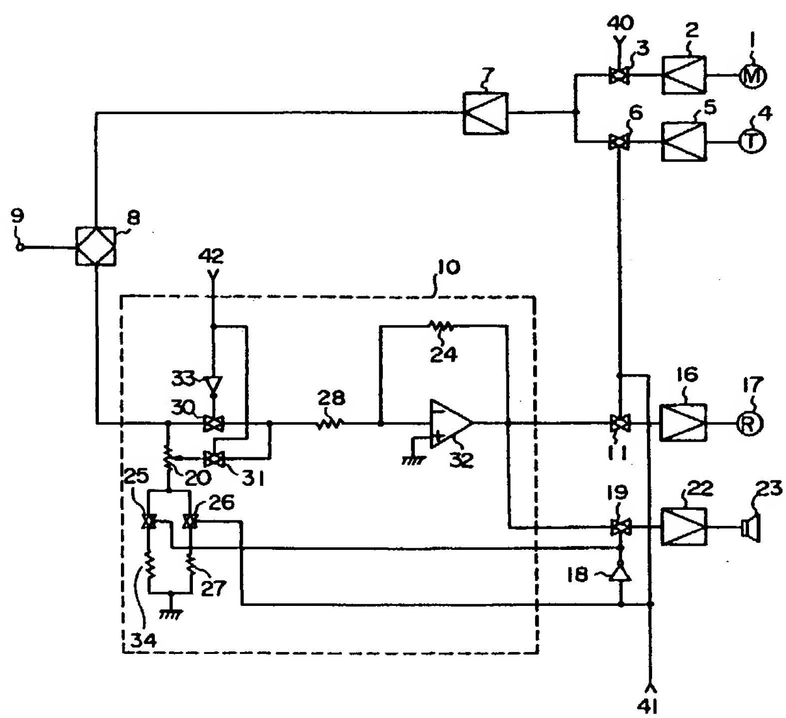
第3図(b)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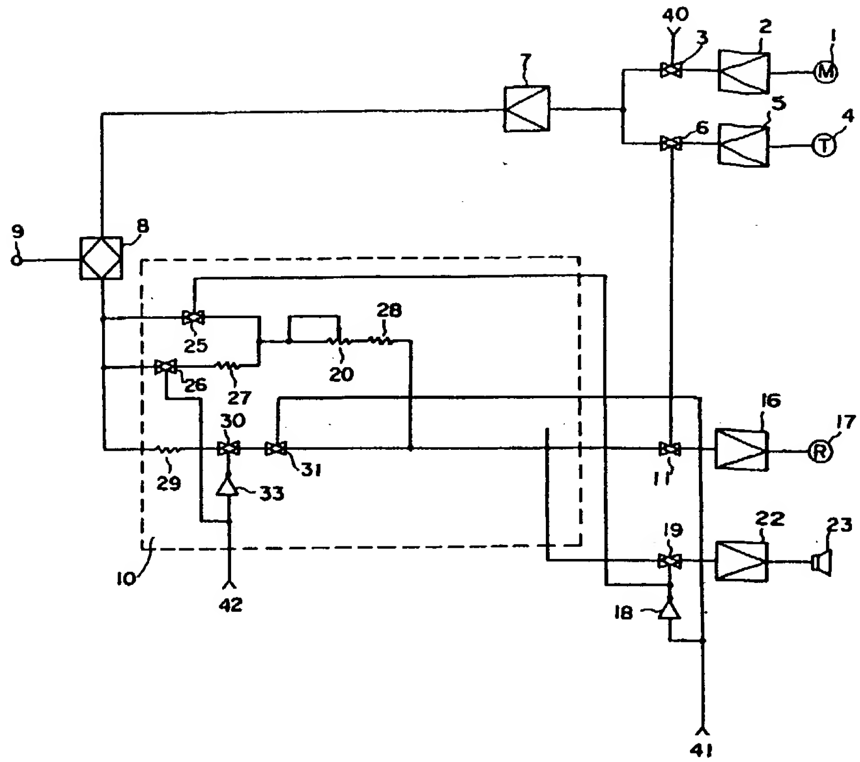
第3図 (c)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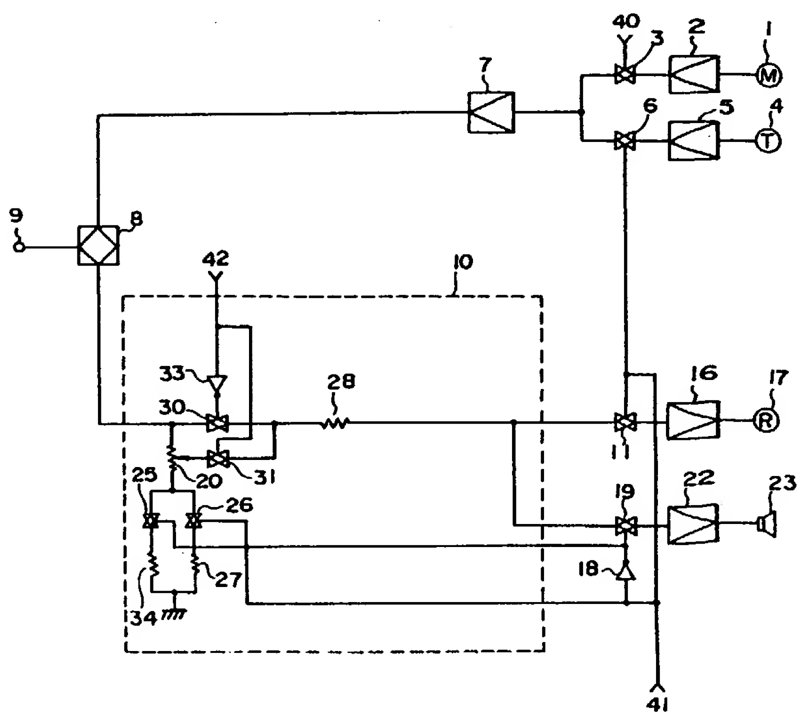
第3図 (d)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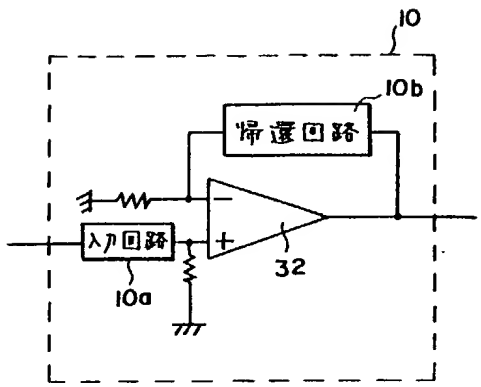
第4図



第1頁の続き
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